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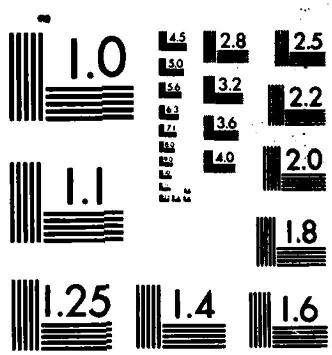

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INTELLIGIBILITY PERFORMANCE OF THE LPC-10 AND APC/SQ SPEECH ALGORITHMS IN A FADING ENVIRONMENT

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Technical Report

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The APC/SQ and LPC-10 Speech Algorithms are evaluated for their performance in a fading channel, listener evaluation is performed for recognition of the Phonetic Alphabet.

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SECTION 1

INTRODUCTION

1.1 INTRODUCTION

This report addresses the effects of fading on two digital speech compression Vocoder algorithms. The two algorithms, Linear Predictive Coding (LPC-10) and Adaptive Predictive Coding with Segment Quantization (APC/SQ) are used as standards by the Department of Defense for use in various military satellite communications systems. Specifically, this study addresses the intelligibility of these algorithms in a fading environment due to high-altitude nuclear events. As a representative example, a variation of a spread spectrum modem referred to as the Transmitted Reference Auxiliary Carrier System (TRACS) was used as the model for the satellite communications link error rate performance.

The intelligibility performance of these voice compression algorithms has been analyzed extensively in both noise free and random error environments. The intelligibility in fading satellite channels is determined by the channel burst error statistics. In turn, the channel burst error patterns depend on the communications modem design, the error correction techniques used and the data interleaver design, as well as the fading statistics and link margin level. A previous phase of this analysis [1] evaluated the behavior of LPC-10 in fading environments, for purposes of developing mitigation concepts for reducing the effect of fading on the output speech. However, the earlier study did not address the intelligibility performance of the algorithm.

Because most of the earlier work concentrated on performance in near ideal conditions, this study needed to use a new approach to more characteristically describe the algorithms' performance in a fading environment. As will be described in this report, the selected approach consisted of using computer simulation models to distort compressed speech data and then re-synthesizing the audio signal for evaluation by human listeners. The major elements of the analysis that was conducted consisted of:

1. Converting and integrating the government furnished LPC-10 and APC/SQ algorithms into existing link simulations.
2. Devising an intelligibility test suitable for evaluating the performance in fading environments.
3. Conducting a series of intelligibility tests in a variety of noise and fading conditions. For LPC-10, both the baseline algorithm and a modified version with mitigation enhancements were tested.
4. Compiling and evaluating the result.

The remainder of Section 1 provides further discusses the background of the study and presents a summary of the key results. Section 2 describes the intelligibility testing concepts and the test tools. Section 3 presents the detailed test results and analysis.

1.2 BACKGROUND

Previous studies [2], [3], [4], [5] have extensively evaluated the performance of LPC-10, APC/SQ and other compressed speech algorithms in both noise free and random

error environments. Also, in a previous phase of this analysis [1], the general performance of LPC-10 was evaluated in a fading and burst error environment. The remaining factor needed for evaluating performance in a nuclear-stressed environment, is the intelligibility of LPC-10 and APC/SQ as a function of the error characteristics on the communications channel.

For a fading or burst error channel, the bit error patterns appear as clusters. This clustering of errors results in either poorer or better intelligibility than that of a channel with randomly distributed errors. At low overall bit error rates, the clustering of errors may cause the loss of a substantial portion of a word, resulting in a listener error, while in a random error channel, the error correction coding in the algorithms would eliminate the errors completely. As the bit error rate increases, the random error channel will reach a point where the errors will destroy the intelligibility of the output speech. However, when the same number of errors are clustered, portions of the speech will be intelligible while others will be totally lost.

In addition to characterizing the performance of LPC-10 as a function of channel signal-to-noise ratio, the previous analysis also developed a set of mitigation concepts for LPC-10. These mitigations concepts, described fully in Reference [1], concentrated on reducing the impact of burst errors on distorting the average shape of individual encoded elements in the compressed speech data. The original five mitigation concepts consisted of:

1. Shortening the time constant in the Error Rate Estimation Algorithm.
2. Updating the error estimation during voiced frames.
3. Adjusting the Pitch Change Limiting factor.
4. Smoothing data over both non-voiced and transition frames.
5. Adding Histogram Smoothing at high error rates.

Of these, all except the Pitch Change Limiting were carried over for evaluation in this analysis. The Pitch Change Limiting concept was eliminated because of its tendency to lock onto high pitch values.

1.3 SUMMARY

The performance analysis of the LPC-10 and APC/SQ algorithms consisted of corrupting compressed speech data with fading-induced errors, synthesizing the audio speech, and then using human listeners to judge the resulting output. Figure 1-1 graphically summarizes the general analysis approach. As shown, the core element of the analysis was the end-to-end computer simulations of the communications link that were configured in MAXIM Technologies Systems Analysis Testbed. Major elements in the simulation included:

1. Government-furnished LPC-10 and APC/SQ analysis and synthesis software.
2. Existing simulations of the DSCS/TRACS system and the nuclear channel fading model.
3. 12-bit A/D and D/A devices for converting audio speech into and out of the computer.

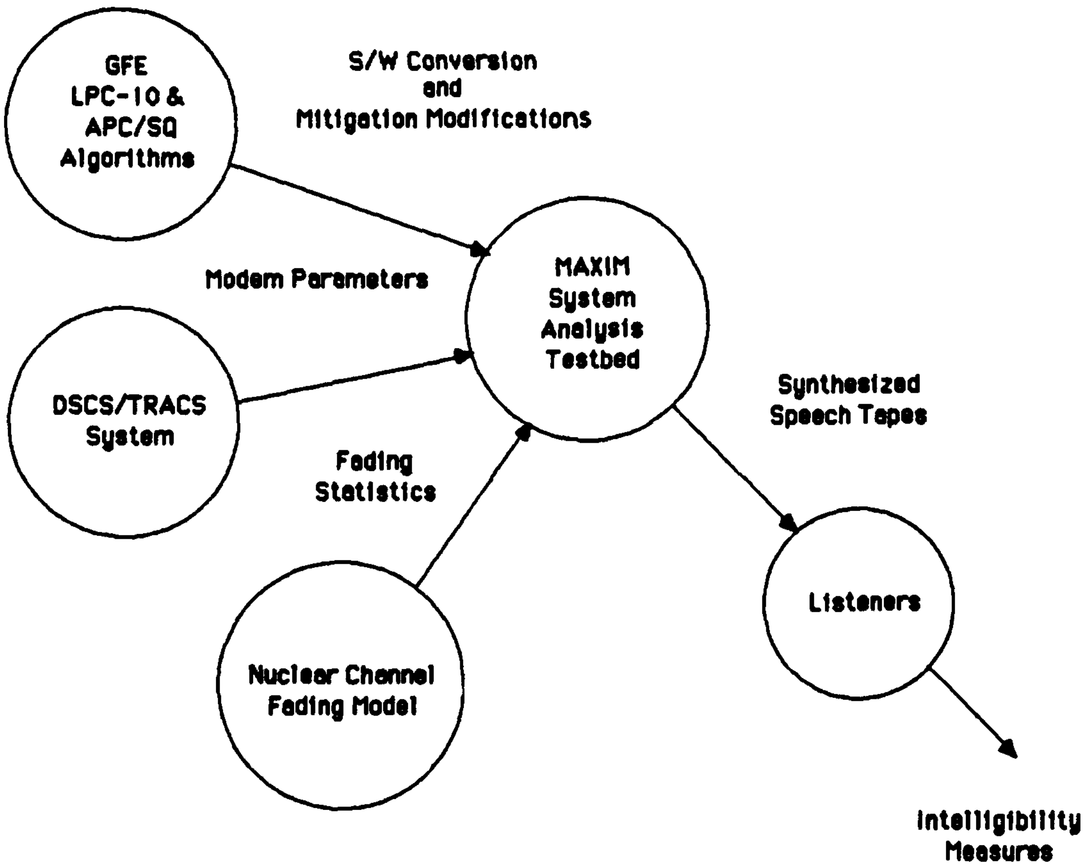

Figure 1-1. Study approach.

Besides the simulation itself, the other critical analysis elements were the human evaluators who listened to the corrupted speech. These people consisted of staff members of MAXIM Technologies. Because they presented a variety of backgrounds, the test results are typical of what would be expected from any random group. Intelligibility scores in early tests were slightly worse than in later ones, so there appears to be some level of learning involved that improves ones ability to decipher a moderate level of distortion. As shown in the following results, the learning curve effect was not very significant, so that the selection of random evaluators is believed to have been a statistically valid approach.

1.3.1 INTELLIGIBILITY CRITERION

For evaluating the performance of speech compression algorithms in a fading and burst error environment, the standard intelligibility scoring tests such as the Diagnostic Rhyme Test (DRT), the Paired Acceptability Rating (PAR), and the Diagnostic Acceptability Measure (DAM) proved to be unacceptable approaches. These tests are designed to evaluate the intelligibility of compressed speech that is already of "good" quality. When used to evaluate compressed speech with errors, the measured intelligibility score rapidly decreases for bit error rates above 1% [5]. For the nuclear-stressed evironment, a different intelligibility criterion was used for the tests described herien. Burst errors in fading links tend to impact entire sylabols or entire words. The DRT tests emphasize the difference between similar monosymbol words, such as "mouse" versus "house". Fading induced errors tend to corrupt the entire word, thus making a test that depends upon one constant meaningless.

The special intelligibility criterion used in this study is based on comprehension levels of random strings of standard military phonetic alphabet words (i.e., ALPHA-ZULU). For descriptive purposes, this approach has been named PACT, the Phonetic Alphabet Comprehension Test. PACT concentrates, not on the clarity of individual sounds as in the DRT or DAM, but on the overall comprehension of a continuous flow of speech. The military alphabet was selected as providing a more typical measure of inteligibility for full duplex communications or voice messages.

Each PACT consists of repeating 104 phonetic words in a random pattern as shown in the example in Table 1-1. The Intelligibility score is defined to be:

$$\text{Intelligibility} = 100 - \frac{\text{Average \# of errors}}{\text{Number of words}}.$$

In a noise free or low-error-rate case, this test yield an intelligibility score of 100%. In less benign environments, the score gradually decreases since an entire word must be obliterated in order to reduce the score by 1 percentage point.

The intelligibility performance varies according to the channel fading characteristics. In a slow fading environment, in which the average fades are significantly longer than the data interleaver length, the errors tend to cluster in bursts roughly equal to the fade duration. This in turn generates

Table 1-1. Sample phonetic alphabet comprehension test.

ED = 1 # WORDS = 26 # POINTS = 104

ST=_ 1_ ERROR TYPE=_ 2_ FADE RATE=_ 0_ PERCENT=_ 3_ MOD=_ 0_

1	ZULU	27	SIERRA	53	BRAVO	79	JULIETT
2	CHARLIE	28	OSCAR	54	ALFA	80	TANGO
3	UNIFORM	29	INDIA	55	SIERRA	81	TANGO
4	NOVEMBER	30	QUEBEC	56	HOTEL	82	OSCAR
5	INDIA	31	ECHO	57	MIKE	83	UNIFORM
6	X-RAY	32	DELTA	58	YANKEE	84	VICTOR
7	KILO	33	BRAVO	59	KILO	85	ROMEO
8	JULIETT	34	X-RAY	60	YANKEE	86	LIMA
9	FOXTROT	35	ZULU	61	UNIFORM	87	UNIFORM
10	VICTOR	36	DELTA	62	PAPA	88	LIMA
11	ALFA	37	VICTOR	63	QUEBEC	89	SIERRA
12	NOVEMBER	38	ECHO	64	BRAVO	90	QUEBEC
13	MIKE	39	GOLF	65	PAPA	91	KILO
14	KILO	40	NOVEMBER	66	YANKEE	92	DELTA
15	FOXTROT	41	ROMEO	67	PAPA	93	ALFA
16	ROMEO	42	JULIETT	68	ECHO	94	NOVEMBER
17	GOLF	43	LIMA	69	ECHO	95	OSCAR
18	HOTEL	44	WHISKEY	70	ALFA	96	GOLF
19	HOTEL	45	GOLF	71	FOXTROT	97	CHARLIE
20	X-RAY	46	CHARLIE	72	X-RAY	98	SIERRA
21	VICTOR	47	OSCAR	73	WHISKEY	99	ZULU
22	FOXTROT	48	ROMEO	74	TANGO	100	LIMA
23	DELTA	49	INDIA	75	MIKE	101	QUEBEC
24	WHISKEY	50	INDIA	76	HOTEL	102	JULIETT
25	MIKE	51	PAPA	77	ZULU	103	YANKEE
26	BRAVO	52	WHISKEY	78	CHARLIE	104	TANGO

alphabet word errors of approximately the same length. Thus, for slow fading, the intelligibility score is approximately equal to 100 minus the average bit error rate. This is a roughly equal to the percent of time that the received signal is above threshold. (Note that this percentage varies directly with the nominal channel signal-to-noise ratio and system fade margin.) For faster fading, the intelligibility performance is a complex function of the channel fading statistics and the specific interleaver and coding equipment used in the system. However, for a given average bit error rate, the intelligibility will always be between the slow fading limit, as a maximum, and the random noise limit, as a minimum.

1.3.2 SUMMARY OF TEST RESULTS

The first, and most extensive, testing was done on the LPC-10 algorithm. The reasons for this were:

1. LPC-10 is vocoder standard being used for secure voice over DSCS satellite links.
2. The algorithm was converted for use in the MAXIM Technologies Testbed during the preceeding analysis.
3. The APC/SQ algorithm was not received from NSA until very late in this analysis.

Three sets of tests were run for evaluating the LPC-10 performance. The first two test sets used the baseline LPC-10 algorithm as it was received from NSA. These two sets served to characterize the nominal performance of the algorithm, and to determine the impact of learning on the validity of the test results. The third set used a modified LPC-10 algorithm

that implemented the mitigation concepts developed in the previous study [1]. Note that these modifications, designed to reduce the observed distortion of the LPC-10 parameters, were based on modifying the speech receiver/synthesizer portion of the algorithm and not the input LPC-10 analyzer or TRACS communications modem.

For each test, a variety of channel bit error rates and fading conditions were used. The results are summarized in Figure 1-2, for the Baseline LPC-10, and Figure 1-3, for the LPC-10 with Mitigation. Figure 1-4 compares the two by overlaying their results. The figures show the Intelligibility Score as a function of the channel Eb/N0 and fading channel decorrelation time. The data labeled "Noise" pertains to a non-fading environment. For the fading environment, the two values of Decorrelation Time shown were selected because they approach the interleaver length used for the TRACS design. The selected values of Eb/N0 shown correspond to average channel bit error rates between 3% at the higher Intelligibility scores and 10% at the low scores. Significant conclusions drawn from the results include:

1. Comparing the pairs of values at equal Eb/N0 values on the same decorrelation time curve, in Figure 1-2, the effects of learning can be seen to be small. The lower values show the results for a completely inexperienced test group while the higher values show the performance on the same test after listening to several other tests.

2. The Intelligibility Score degrades rapidly in both fading and non-fading environments, changing from near 95% to 75% for less than a 3 dB reduction in Eb/N0. This corresponds to an increase in average bit error rate from 3% to 10%.

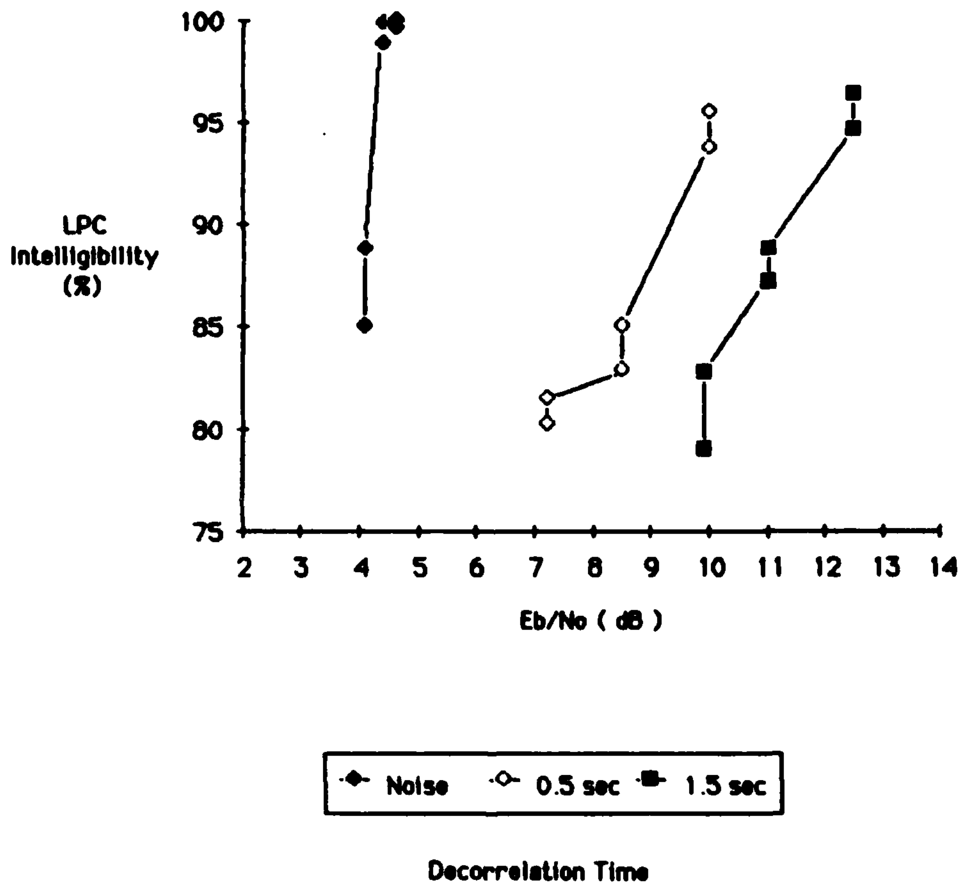

Figure 1-2. Baseline LPC-10 intelligibility performance.

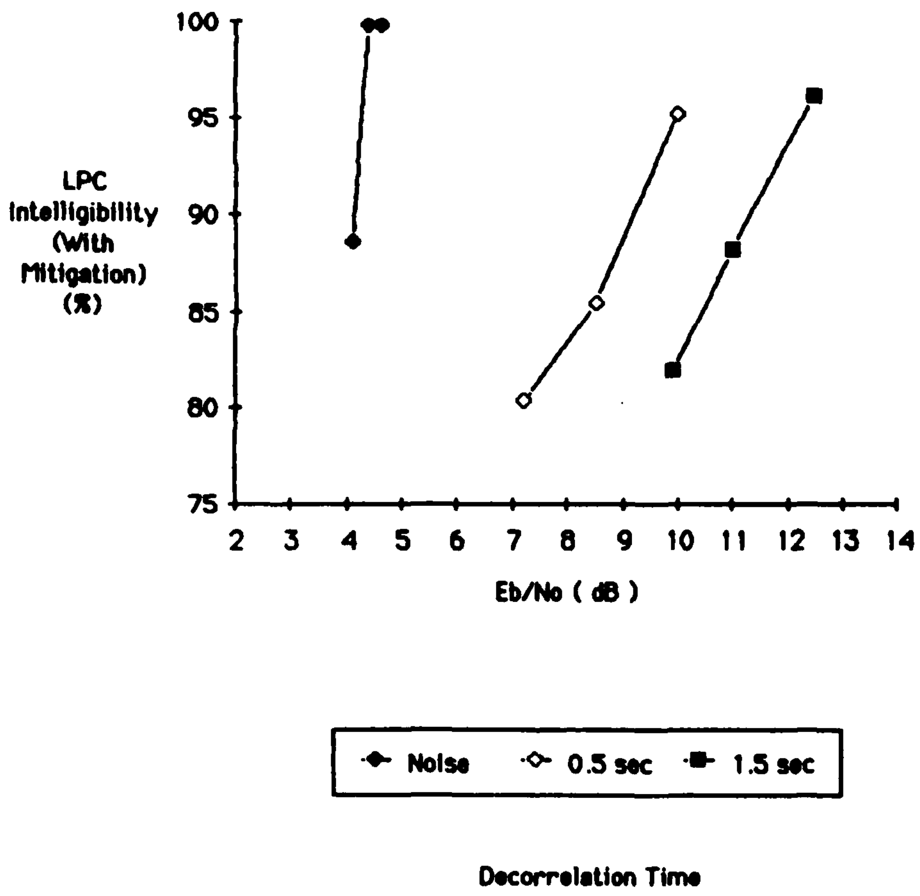

Figure 1-3. Modified LPC intelligibility performance.

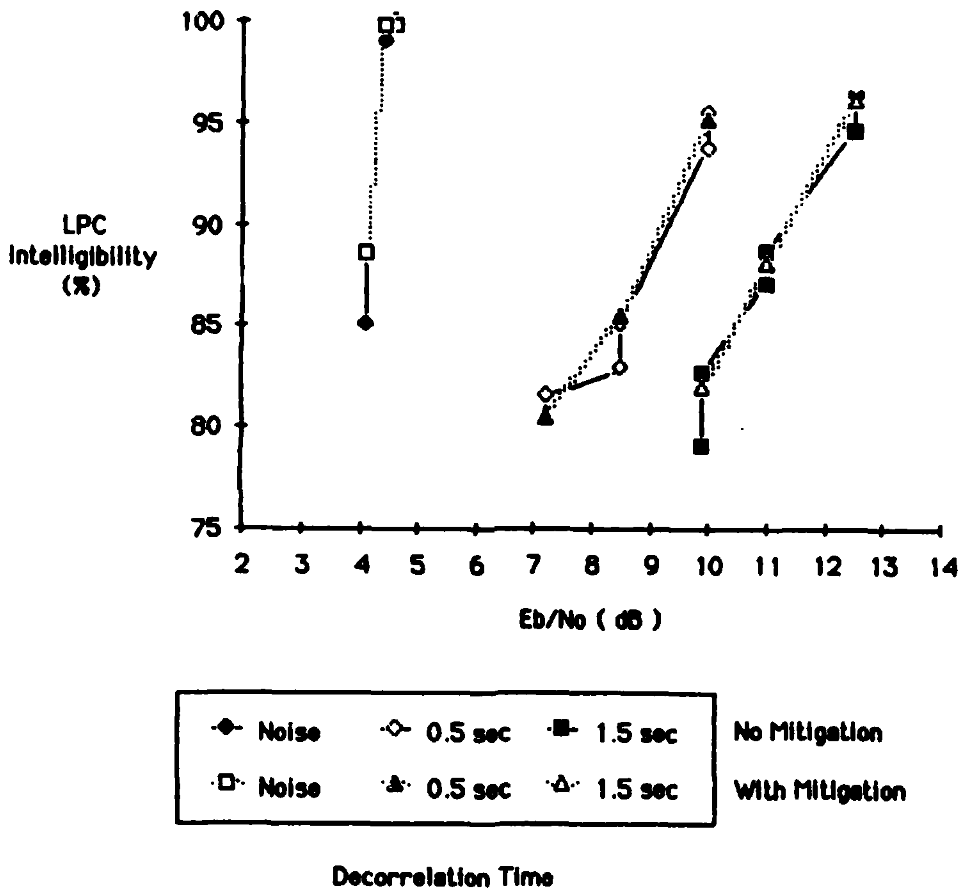

Figure 1-4. Comparison of LPC performance with and without mitigation modifications

3. Comparing the Baseline results to those with the Mitigation enhancements, as shown in Figure 1-4, it can be seen that the current mitigation concepts are not very effective. This is due to the bursty nature of the output errors from the convolutional decoder and interleaver in the TRACS implementation. Since the errors occur in dense bursts, the LPC-10 parameters are so corrupted that there is little possibility of accurately recovering the intended speech pattern by modifying the LPC-10 algorithm.

Since the APC/SQ algorithm was not operational in the Testbed until late in the study, only a single set of tests were evaluated. These tests followed the same pattern as the LPC-10 tests in that they were run with a similar set of noise and fading conditions. The fading decorrelation times evaluated are faster and proportional to the higher data rate (9.6 KBPS versus 2.4 KBPS) of APC/SQ. Mitigation concepts were not investigated for the APC algorithm. Figure 1-5 summarizes the APC/SQ test results. The results are similar to the LPC-10 results in that the intelligibility falls off at lower Eb/N0 values and degrades roughly in parrallel with the channel Bit Error Rate.

Because the LPC-10 and APC/SQ tests were run with different decorrelation times and interleaver lengths, comparison of the results must be done on a relative basis. Figure 1-6 shows the intelligibility performance of the two algorithms as a function of the ratio between

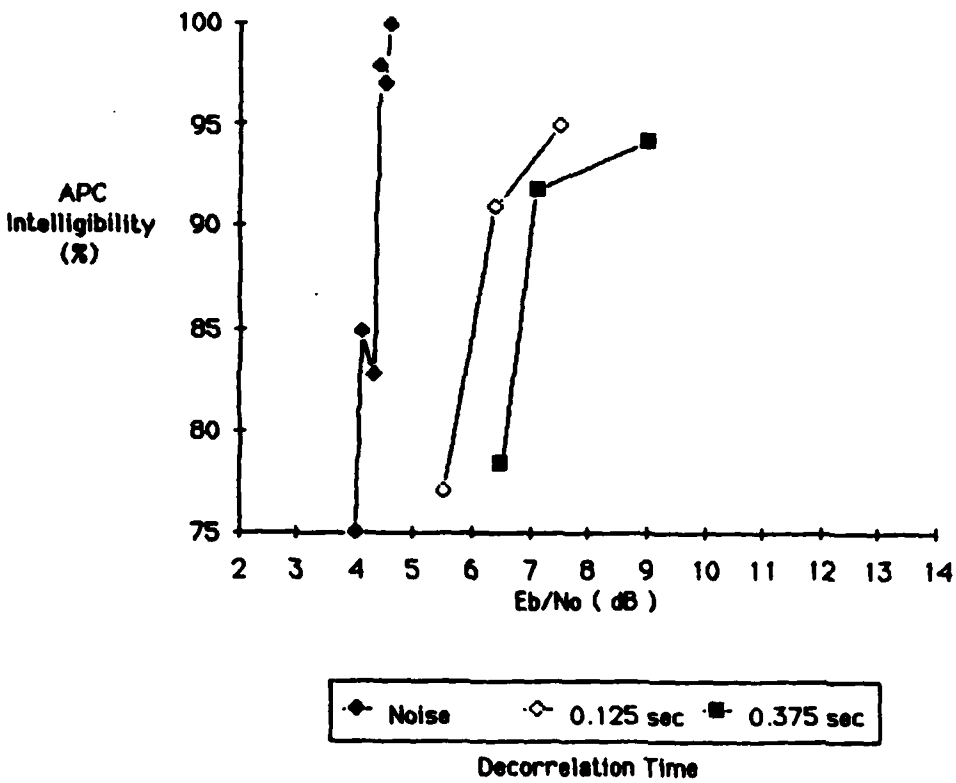

Figure 1-5. APC/SQ intelligibility performance.

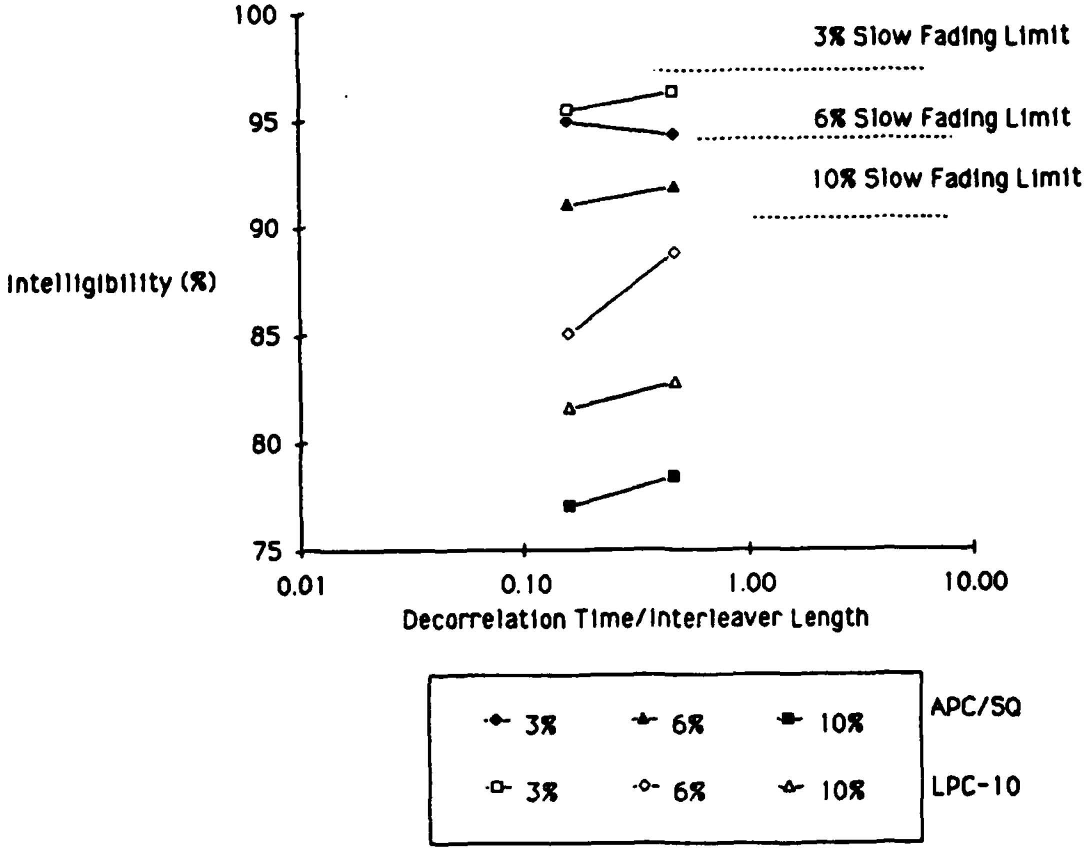

Figure 1-6. Comparison LPC-10 and APC/SQ intelligibility using relative decorrelation time.

the decorrelation time and the data interleaver length. The results show that LPC-10 performs slightly better at low error rates (<3%) and significantly better at high error rates (~10%), but that APC/SQ performs better at moderate error rates (~6%). The better performance of LPC-10 is due to the roughly equal error correction capabilities of the two algorithms at low error rates, and the short "memory span" of LPC-10 at the high error rates. At moderate BER, APC/SQ outperforms LPC-10 because the higher data rate allows more bit errors to be hidden in the synthesized output speech as noise or other spurious sounds.

Section 3 presents the results of all of the tests in more detail and in other formats, including results as a function of Decorrelation Time, and of Bit Error Rate. Since the DSCS/TRACS channel was used as the model for all the tests, these results may not be completely valid for other links. In particular, the results can vary considerably with data interleaver size and with specific modulator/demodulator characteristics. However, the general trends and nature of the intelligibility performance is expected to be valid for most other standard military satellite communications links such as MILSTAR for comparable interleaver span-to-decorrelation time ratios.

1.4 CONCLUSIONS

The test sets performed have served to characterize the intelligibility performance of the LPC-10 and APC/SQ speech compression algorithms in nuclear-stressed fading environments. The intelligibility testing was based on the Phonetic Alphabet Comprehension Test (PACT) which was developed as part of the study. PACT was developed and used

in order to obtain a better measure of overall comprehension under the marginal performance expected in a nuclear environment than that provided by the more conventional DRT or DAM testa.

From the summary of results in Section 1.3 and the detailed test results in Section 3, some conclusions can be drawn from the study:

1. LPC-10 and APC/SQ provide comparable intelligibility performance at low BER levels.

2. LPC-10 provides better intelligibility at high BER (>10%) while APC/SQ performs better at moderate BER (6%).

3. Mitigation concepts based on smoothing individual coefficient statistics are not substantially effective in fading environments.

4. For a constant average BER, speech compression algorithms yield a higher intelligibility with burst errors rather than random errors. This results from the fact that speech also comes in bursts, so that a burst of errors causes loss of only a word or so while randomly distributed errors will distort a greater percentage of the overall conversation.

Interleavers can be used to randomize burst error patterns. However, the value of interleavers for improving compressed speech performance has both positive and negative aspects. Typical decoding algorithms operate best when the

input error statistics are random. Because of this, output BER from the decoder can be minimized by using the interleaver to randomize the burst errors due to the channel. As a rule of thumb, an interleaver delay of 10 times the channel decorrelation time is required to achieve a random error pattern for input to the decoder. However, this would result in a 10 to 20 second delay in speech interactions between two talkers. Because of this, shorter interleavers were assumed for use in this study. For both algorithms, a 16384 point random interleaver was used. This yields a memory size of 3.4 seconds for LPC-10 and 0.85 seconds for APC/SQ. This delay is probably unacceptable for full duplex speech, but may be acceptable for simplex voice circuits.

To attempt to obtain a general characterization of algorithm performance for varying interleaver length, the intelligibility performance was plotted versus a Relative Decorrelation Time. While the results shown in Figure 1-6 give some insight into the impact of varying the interleaver length, the tests do not cover interleaver lengths shorter than the decorrelation time. While it is expected that the intelligibility would smoothly approach the noise-only and slow-fading asymptotes for varying interleaver lengths, additional testing would be required to fully quantify the performance of the algorithms for short interleavers. Other additional evaluations and studies that would be of value include:

1. Further analysis of the proposed mitigation concepts to determine why they did not improve intelligibility, even though they appeared to improve the quality of the output data fed into the LPC-10 synthesizer.

2. Further analysis of the statistical distribution of the phonetic word errors and their relationship to the channel burst error statistics.

3. Evaluation of speech intelligibility performance for non-encoded communications channels, so that fading effects can be separated from decoder and interleaver effects.

4. Evaluation of other mitigation concepts for both LPC-10 and APC/SQ that would be more effective in strong fading conditions. Pattern recognition techniques are one potential candidate which may be more suited for recovering the content of severely corrupted LPC-10, or APC/SQ, reflection coefficients and other parameters. Smoothing techniques, such as used in the current mitigation concepts, can only eliminate moderate distortions. Since speech data has only a finite set of phonemes, pattern matching with a fixed set of data and phoneme substitution for severely distorted data may be more useful.

SECTION 2

PERFORMANCE EVALUATION TECHNIQUES

2.1 GENERAL APPROACH

This section describes the general approach used to evaluate the performance of the LPC-10 and APC/SQ speech compression algorithms in a fading and burst noise environment. Brief overviews of the algorithms are presented in Appendices A and B.

2.2 PHONETIC ALPHABET COMPREHENSION TEST

In earlier work [2],[3],[4] intelligibility measures such as the Dynamic Rhyme Test (DRT), the PAR and the DAM have been used to characterize the intelligibility of speech compression algorithms in error free and low-error-rate environments. These measure are adequate for characterizing the performance in near ideal conditions, but are not adequate for evaluating intelligibility in marginal conditions such as might be encountered on a satellite communications link in a nuclear-stressed environment. In such conditions, the bit error rate may be only marginally acceptable, far worse than the 1% limit [4] for acceptable LPC-10 performance. Because of this a new intelligibility measurement concept was required for this study.

The evaluation criterion selected for the intelligibilty test is known as the Phonetic Alphabet Comprehension Test (PACT). In this test, listeners are presented with a

continous sequence of military phonetic words (e.g., ALPHA, BRAVO, CHARLIE, ... ZULU) that have passed through a fading communications channel. Each test consists of 104 words made up of the 26 alphabet words repeated 4 times in a random order. The value of the test is that it concentrates on the overall comprehension of the message, rather than on the clarity and distinction of individual sounds (such as distinguishing between moot and boot in the DRT).

The intelligibility score was determined by counting the total number of incorrect words for each listener. Allowances were made in the scoring to ignore missed or extra words. This is to correct for situations when the listener loses synchronization with the data due to spurious sounds interpreted as separate words or word dropouts by the listeners. With these corrections, the resulting errors were averaged over the set of listeners to determine an average word error rate for the test. From this, the Intelligibility Score is given by:

$$\text{Intelligibility} = 100 - \frac{\text{Average \# of Errors}}{104} .$$

A sample test score sheet is show in Table 2-1 to demonstrate the scoring technique.

For reference, the tests were first run with no channel errors. As expected, these preliminary tests resulted in 100% intelligibility scores for both LPC-10 and APC/SQ. This was by design, so that the variability in intelligibility would occur at the moderate bit error rates expected on the satellite links rather than at low rates used for evaluating

Table 2-1. Sample PACT test score sheet.

SEED = 2 # WORDS = 26 # POINTS = 1Ø4

TEST=_ 2_ ERROR TYPE=_ 2_ FADE RATE=_ Ø_ PERCENT=_6_ MOD=_ Ø_

1	Z	27	Y	53	N	79	
2	R	28	I	54	M	8Ø	
3	T	29	K	55	W	81	X
4	W	3Ø	Y	56	B	82	V
5	L	31		57	M	83	Z
6	K	32		58	H	84	Y
7	N	33	X	59	J	85	F
8	U	34	V	6Ø	O	86	
9	R	35	L	61	A	87	E
1Ø	S	36	C	62	K	88	Q
11	C	37	I	63	G	89	G
12	L	38	J	64	F	9Ø	P
13	S	39	S	65	Y	91	Q
14	G	4Ø		66	E	92	I
15	B	41	M	67	P	93	R
16	Q	42	X	68	N	94	G
17	E	43	R	69	O	95	U
18	O	44	B	7Ø	F	96	W
19	A	45	J	71	C	97	P
2Ø	H	46	X	72	D	98	U
21	F	47	N	73	J	99	
22	Q	48	3	74	I	1ØØ	
23	O	49	Z	75	K	1Ø1	
24	T	5Ø	T	76	D	1Ø2	
25	A	51	~~5~~ P #52	77		1Ø3	
26	V	52	B	78			

16 errors

the basic algorithms. The error free tests also served as training for the evaluators, so that they were comfortable with the nature of the basic compressed speech output.

2.3 ANALYSIS EQUIPMENT AND SOFTWARE

The equipment and software used to generate the corrupted compressed speech was configured in MAXIM Technologies' System Analysis Testbed. The overall structure of the simulator used in this study is shown in Figure 2-1. Permanent equipment in the testbed that was used in this study consisted of:

1. VAX 11/750 computer and peripherals.
2. 12-bit A/D and D/A for data acquisition.
3. Nuclear channel models.
4. Modem and receiver simulation software.

In addition, software for the LPC-10 and APC/SQ algorithms was converted and integrated into the Testbed. This software was provided to MAXIM Technologies by NSA [8], [9] in a generic format which was then converted for use on the VAX 11/750. A brief overview of these algorithms and the remainder of the analysis software is contained in Appendices A and B.

The model used for the satellite communications link was the Transmitted Reference Auxiliary Channel System (TRACS) modem. This provides a generic example of a PSK single channel communications signal as well as typical block interleaving and Viterbi decoding schemes. Since all the tests used this model, the results are only completely valid for LPC-10 and APC/SQ as used on TRACS. However, since TRACS

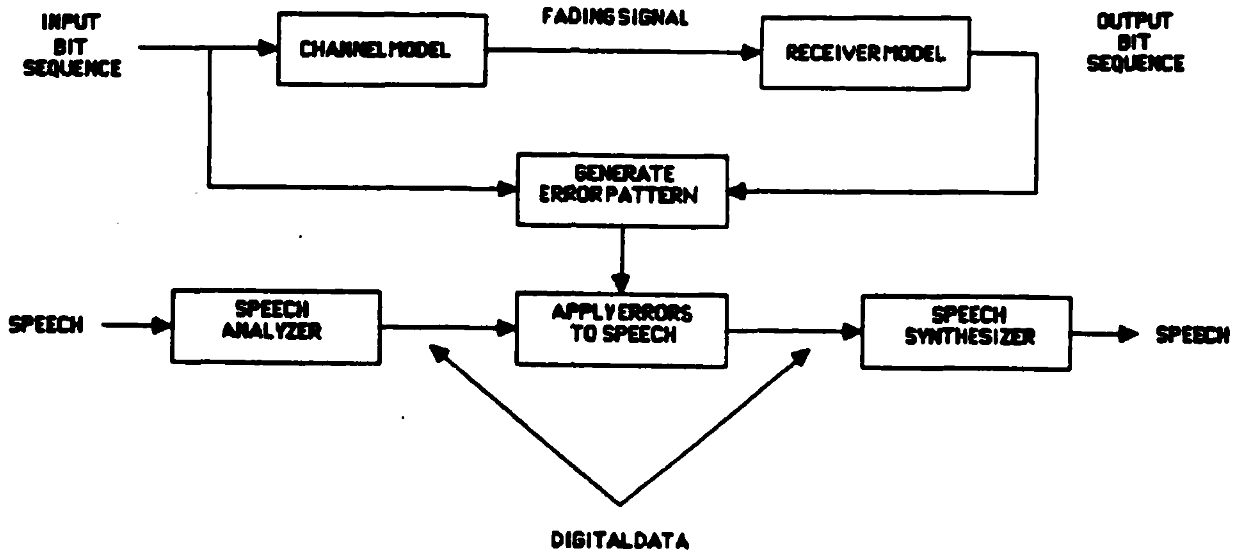

Figure 2-1. System analysis testbed configuration for PACT testing.

system is somewhat representative of other military digital speech communications systems, the results are believed to be relatively characteristic of most other systems with similar interleaver delay. A brief overview of the TRACS receiver is also included in Appendix A.

2.4 TEST CONDITIONS

The intelligibility tests were run for a variety of noise and fading conditions that were selected to characteristically match the characteristics of the TRACS modem and to span the range of channel characteristics expected in a moderate nuclear environment. Since tests involving human evaluators are both time consuming and relatively expensive, a fixed set of a priori conditions were selected and then used for evaluating both algorithms.

For noise only tests, the modem/channel simulator was iterated until a set of error patterns were obtained yielding 0.3, 1.3, and 5.1 % average bit error rates. Then, with the addition of the fading channel model, additional error pattern files were obtained for 3, 6, and 10 % average bit error rates with fading decorrelation times of 0.5 and 1.5 seconds. These two steps resulted in a set of nine error pattern files that were then used repeatedly for each of the four sets of tests:

1. Baseline LPC-10
2. Baseline LPC-10 retake (for learning curve effects)
3. LPC-10 with mitigation enhancements
4. Baseline APC/SQ.

The same error patterns and word sequences were used for each of the four tests. By doing this, the effects of identical error bursts could be observed for LPC-10 both with and without the mitigation modifications.

SECTION 3

INTELLIGIBILITY TEST RESULTS

3.1 OVERVIEW

This section presents a more detailed description of the results of the PACT intelligibility tests to complement the overview presented in Section 1. The results can be viewed in several formats, with each alternative providing a different perspective on the behavior of the speech compression algorithms in fading environments. Table 3-1 lists the three formats used in the study and summarizes the value of each in evaluating the performance of the algorithms. Sections 3-2 presents the test results for the LPC-10 algorithm. Section 3-3 presents the APC/SQ test results and compares its performance with the LPC-10 algorithm.

3.2 LPC-10 TESTS

Testing of the LPC-10 algorithm served three major purposes:

1. Defining the basic characteristics of compressed speech and requrements for designing an effective intelligibility test.

2. Setting a baseline against which the mitigation enhancements to LPC-10, and the baseline APC/SQ, could be compared.

Table 3-1. Alternatives for assessing intelligibility performance results.

INTELLIGIBILITY VERSUS	PARAMETRIC IN	CHARACTERISTICS TO OBSERVE
$\frac{Eb}{No}$	Decorrelation Time	Shows specific performance for DSCS/TRACS receiver. Slower fading requires higher Eb/No than fast fading.
Average Bit Error Rate	Decorrelation Time	More generic since RF receiver effects are eliminated. Coder and Interleaver yield characteristic burst error patterns. Intelligibility relatively insensitive to decorrelation time.
Decorrelation Time	Average Bit Error Rate	Shows that intelligibility varies between noise limit, as a minimum, and slow fading, as a maximum, for a constant BER. Transition region occurs approximately where decorrelation time equals interleaver length.

3. And finally, the basic purpose, demonstrating the performance of the LPC-10 algorithm in a fading environment.

The results of the first test set, using the baseline LPC-10 algorithm and the TRACS receiver model, are shown in Figures 3-1 through 3-3. Looking first at Figure 3-1, which shows the performance for the TRACS receiver as a function of Eb/N0, three features can be noted:

1. Intelligibility falls off more rapidly, in terms of Eb/N0, at faster fading rates (i.e., lower decorrelation times.) This results from the parrallel fall off of BER for fast fading.

2. Required Eb/N0, for a constant intelligibility score, increases as the decorrelation time increases.

3. For the TRACS receiver, a change of 3 dB in Eb/N0 can change the intelligibility from 80% to close to 100%.

For reference, with an intelligibility factor of 80% (Implying that one of five words is garbled.), a conversation would most likely be rated unacceppable. Speech at this level requires considerable patience and "integration" by the listener to extract the content of the message. On the other hand, an intelligibility score of 95% is comparable to a standard telephone conversation.

The intelligibility of LPC-10 changes approximately in parallel with the average bit error rate of the digital data

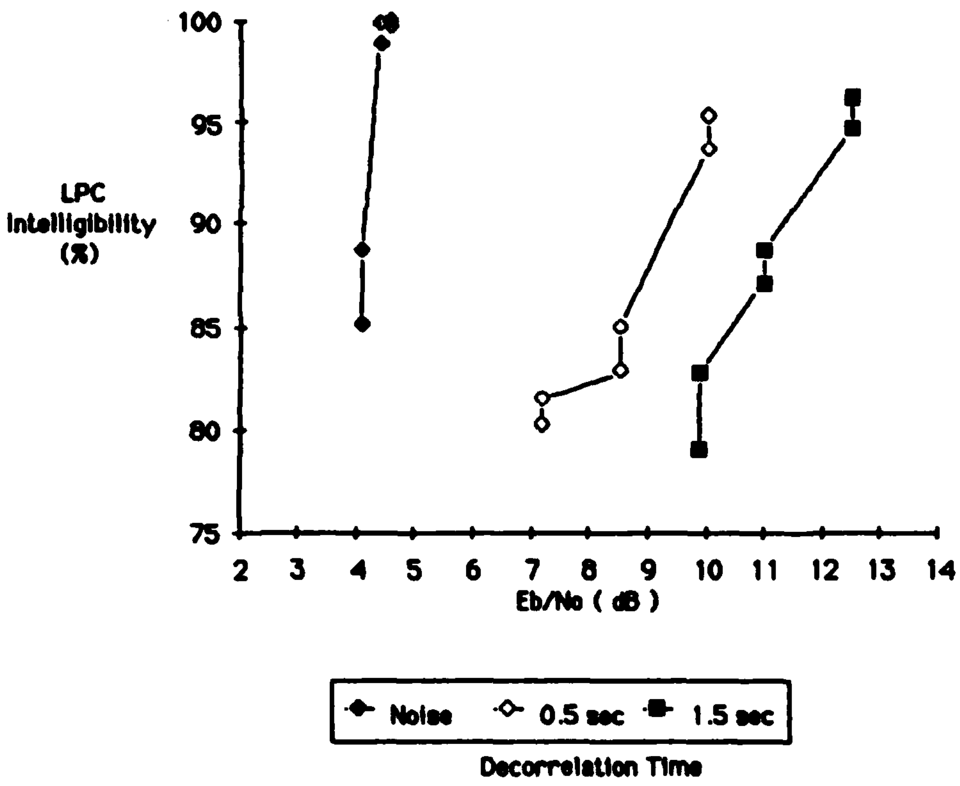

Figure 3-1. LPC-10 intelligibility performance for the TRACS receiver.

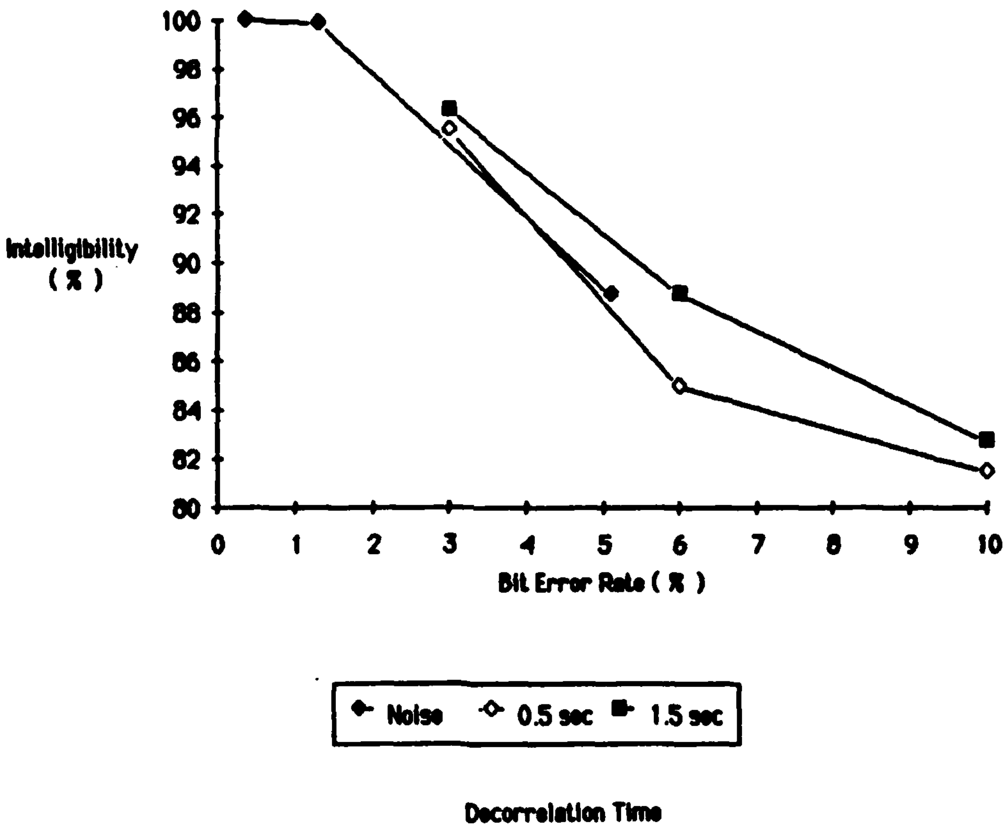

Figure 3-2. Baseline LPC-10 intelligibility performance versus average BER.

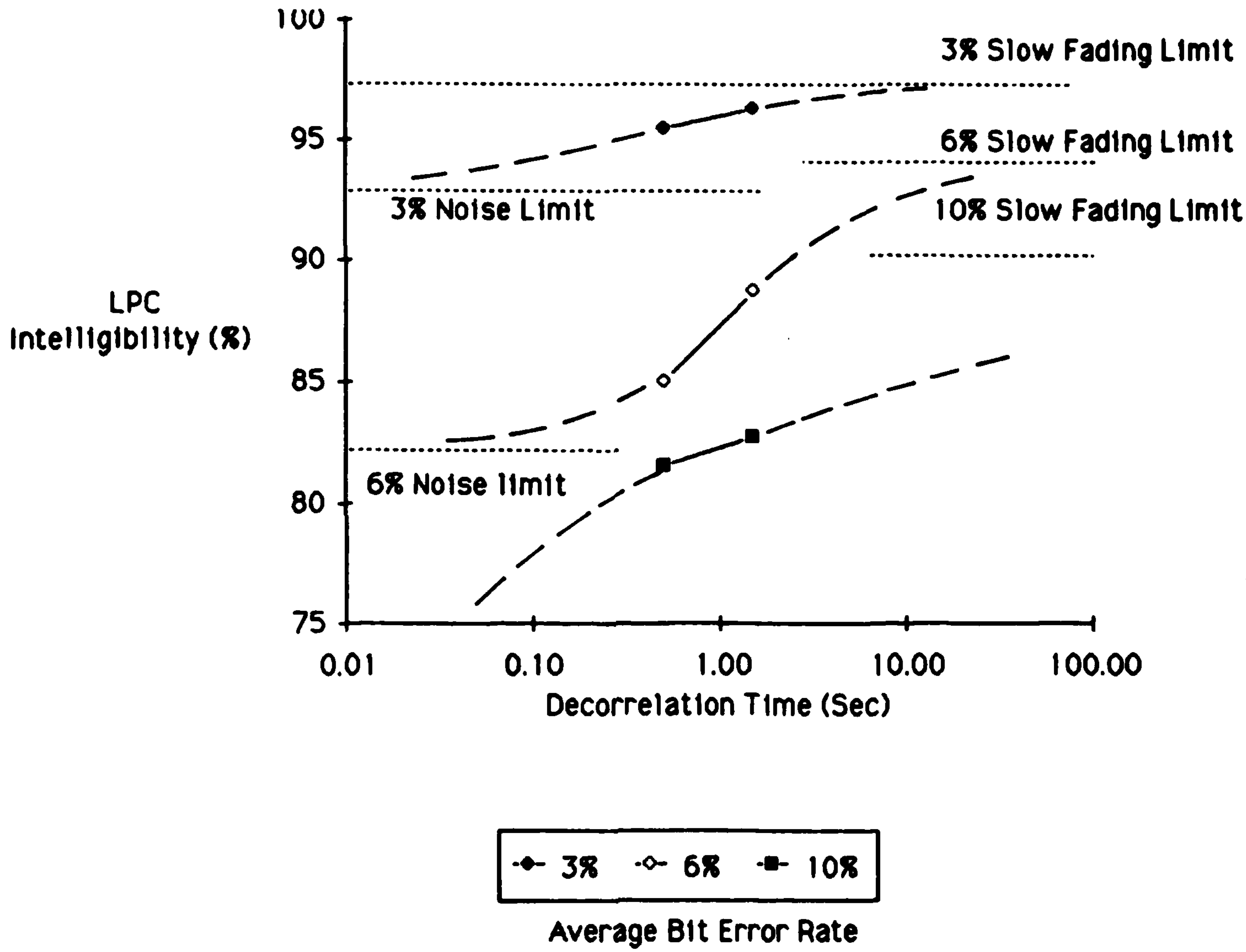

Figure 3-3. Baseline LPC-10 intelligibility performance versus decorrelation time.

input to the output synthesizer. For the TRACS receiver, in both fading and non-fading burst error environments, the PACT intelligibility measure falls below 95% at a BER of approximately 3% and below 80% at a BER of approximately 10%.

Since the results shown in Figure 3-1 assumed the use of the standard TRACS receiver, the translation of Eb/N0 to Intelligibility is only completely valid for the TRACS channel. In order to extrapolate the performance of LPC-10 to other communications systems, the specific effects due to the TRACS demodulator must be eliminated. One simple approach for accomplishing this, that preserves the burst error effects from the fading channel, is to plot the performance versus Average Bit Error Rate (BER) instead of Eb/N0. By doing this, and assuming that the front-end of the TRACS demodulator does not impact the burst error patterns generated by either the interleaver/decoder or the fading channel, the test results essentially become generic measures of the intelligibility performance of the algorithm. Figure 3-2 shows this generic representation of the performance using the same data as used for the TRACS receiver tests. Features to note in Figure 3-2 include:

1. The Intelligibility falls off at a BER of approximately 2%, roughly independent of the decorrelation time, T0.
2. Variations with T0 are minor for the values tested. This probably results from the selection of T0 values close to the interleaver duration.
3. An approximate linear fit to the results shows a 2-to-1 slope, i.e.

$$\text{Intelligibility} = 100 - 2 \times \text{BER} (\%) .$$

A final approach to view the results is shown in Figure 3-3. Here, the parameters of Figure 3-2 have been reversed so that the decorrelation varies along the x-axis and the curves are parametric in BER. From these curves, the variation of intelligibility between the slow-fading and noise-only limits becomes apparent. The lower noise-only limit characterizes performance when the duration of the error patterns is much less than the interleaver length and decoder memory. In these cases, the resulting errors in the LPC-10 data are close to randomly distributed. With the average number of bit errors held constant, the near-random distribution of errors distorts a greater number of speech words, hence reducing the overall intelligibility. At the other extreme, the slow-fading limit, the input burst error patterns are approximately equal to the fade duration. For these tests the data interleaver was selected to be 16384 samples, or 3.4 seconds, in length. With the assumption that the data interleaver is shorter than the burst error pattern, the output bit errors fed into the LPC-10 synthesizer will also appear as a burst dropout of approximately equal length.

With this, the defined intelligibility measure for a constant average BER will be equal to:

$$\text{Intelligibility}\Big|_{\text{slow fade}} = 100 - \text{BER}\ (\%)\ .$$

Note that this limit exceeds the trend of the results shown in Figure 3-2, so that the results in Figure 3-2 are not completely slow fading situations.

For simplicity, the preceding discussion covered only the Baseline LPC-10 algorithm and not the modified algorithm with the proposed mitigation enhancements. As shown in the summary results in Section 1, the tests with the mitigation enhancements were somewhat discouraging because of the lack of any appreciable difference between the intelligibility performance of the two versions of the algorithms. As another example of this, Figure 3-4 shows the results for the modified algorithm as a function of BER with decorrelation time as a parameter. Comparing these results with Figure 3-2, again shows only minor differences between the two.

3.3 APC TEST RESULTS

Results of the APC/SQ PACT tests are shown in Figures 3-5 through 3-7. Since APC/SQ uses a higher data rate, 9600 BPS versus 2400 BPS for LPC-10, the tests were run with decorrelation times scaled by the data rate ratio. Also, the data interleaver length scales so that the effective length in the APC/SQ tests was approximately 0.8 seconds.

Keeping in mind that the main purpose of the APC/SQ tests were to compare performance with the lower data rate LPC-10 algorithm, the results of the APC/SQ tests should be viewed in parallel with the LPC-10 results. In Figure 3-5, the same characteristic fall-off in intelligibility occurs as Eb/N0 decreases as was shown in Figure 3-2. The noise only results appear nearly identical in terms of Eb/N0, however, this is due to the fact that, for the TRACS receiver, BER changes several orders of magnitude for less than a 1 dB variation in Eb/N0. Comparing Figures 3-2 and 3-6, LPC-10 actually performs slightly better than APC/SQ for noise

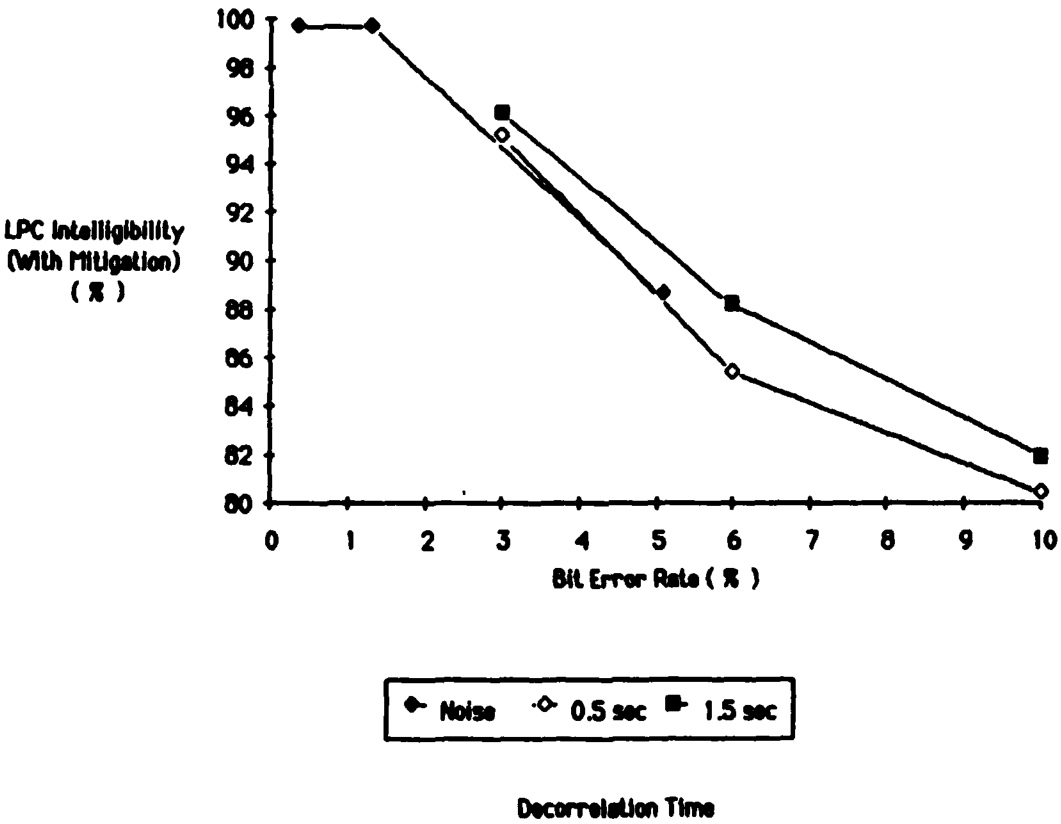

Figure 3-4. Modified LPC-10 intelligibility performance versus average BER.

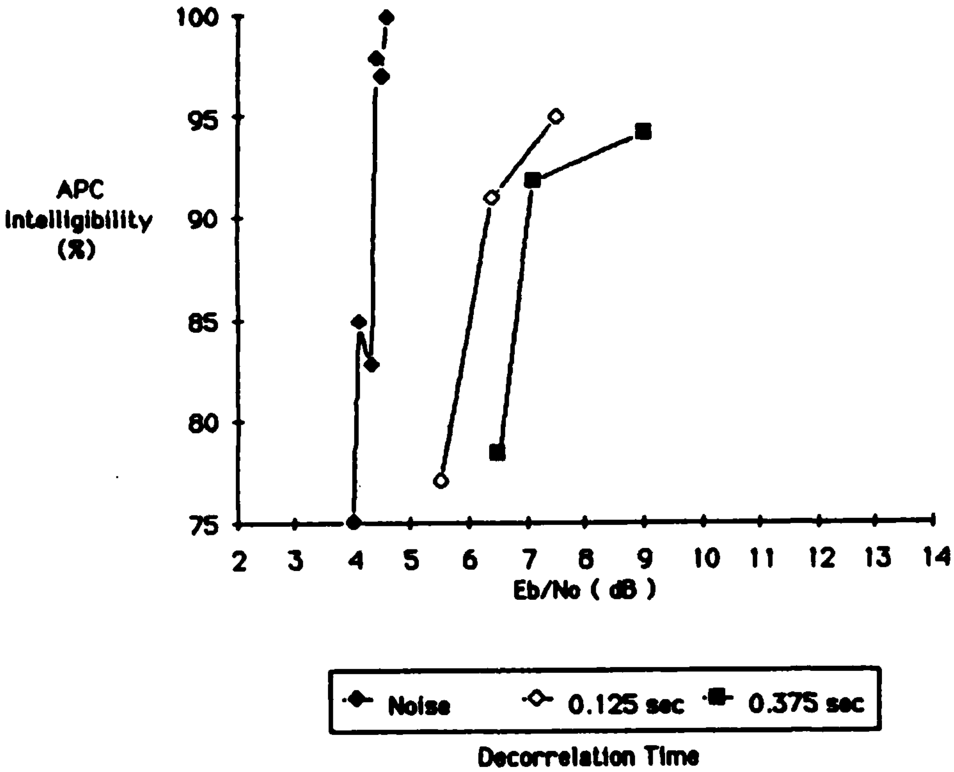

Figure 3-5. APC/SQ intelligibility performance for the TRACS receiver.

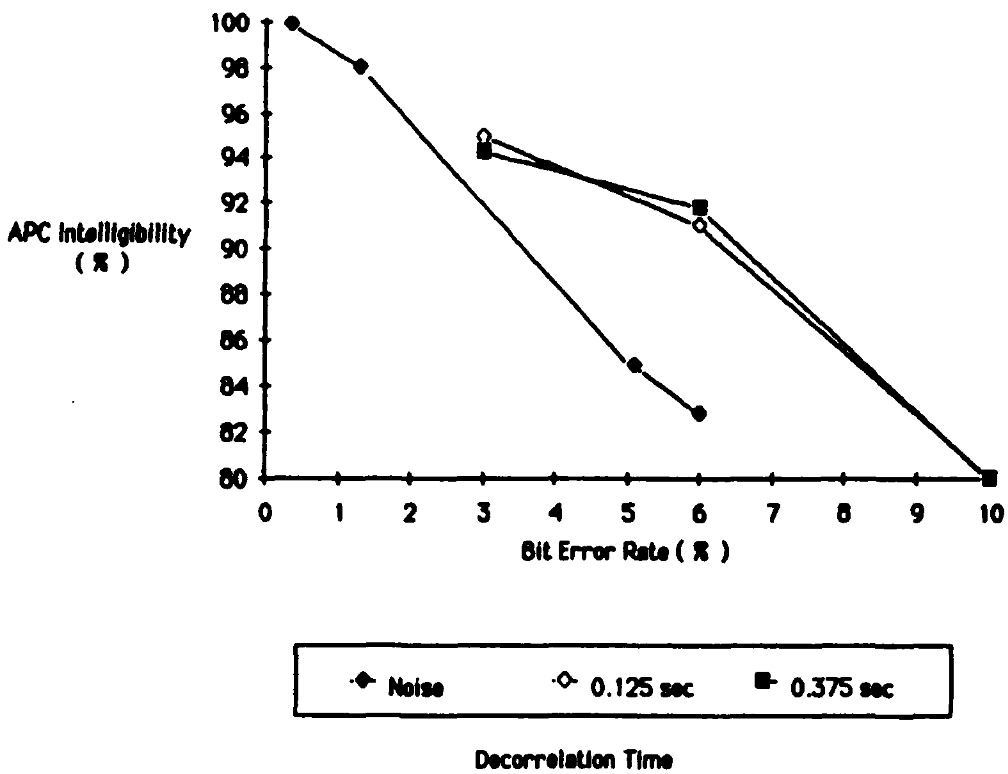

Figure 3-6. APC/SQ intelligibility performance versus average BER.

only conditions. Figure 3-6 also shows that APC/SQ intelligibility performance is, like LPC-10, relatively insensitive to decorrelation time, for constant BER. Again, this characteristic only occurs because the selected decorrelation times are close to the interleaver length. Note that the APC/SQ intelligibility falls off at a faster rate than LPC-10 for BER above 6%.

The results for the fading tests, showing again the asymptotic behavior for noise and slow-fading, are shown in Figure 3-7. Since these tests assumed faster decorrelations times than were used in the LPC-10 tests, it is difficult to directly compare the two sets of results. However, by converting the decorrelation time to a dimensionless factor scaled by the interleaver length,

$$\text{Relative Decorrelation Time} = \frac{\text{Decorrelation Time}}{\text{Interleaver Length}}$$

some approximate comparisons can be made. These results are shown in Figure 3-8. The graph shows that APC/SQ performs better than LPC-10 at 6% BER but worse than LPC-10 at 3% and 10%. This effect occurs because:

1. At low error rates (<3%) the performance is nearly equal.

2. At moderate error rates (6%), APC/SQ performs better than LPC-10 because at the higher BER the errors are distributed enough that they tend to distort rather than obliterate words.

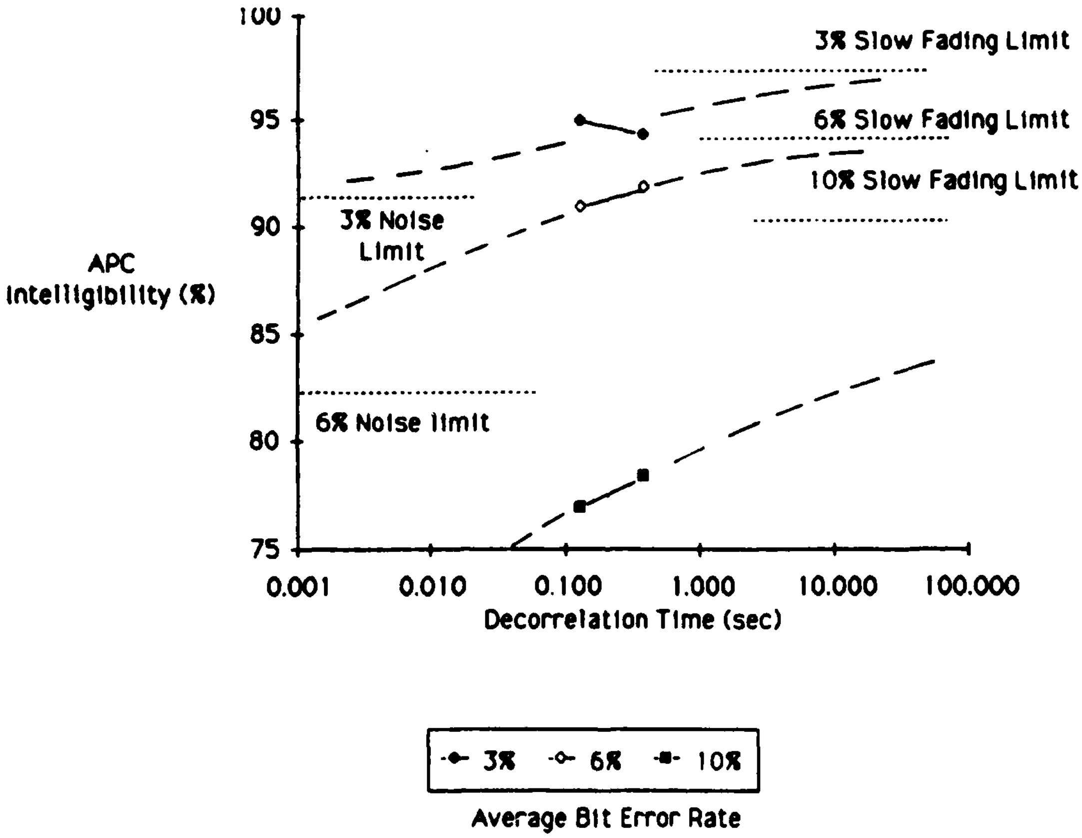

Figure 3-7. APC/SQ intelligibility performance versus decorrelation time.

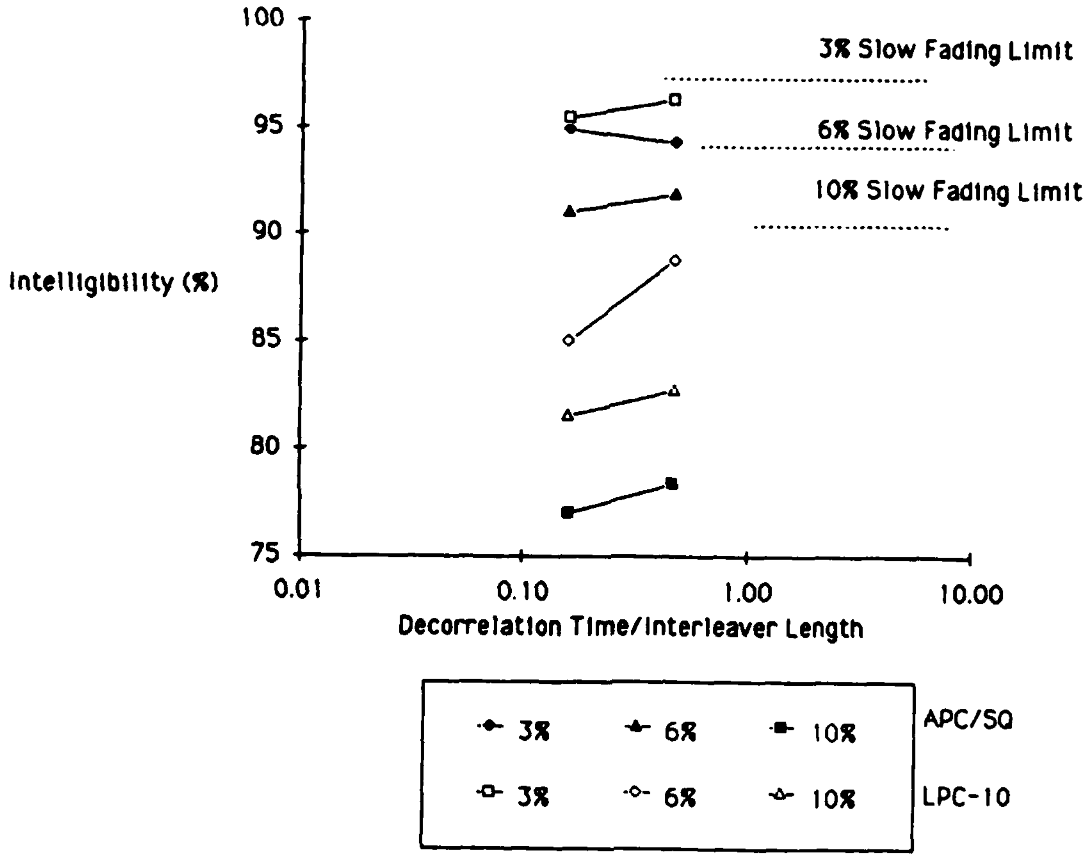

Figure 3-8. Comparison LPC-10 and APC/SQ intelligibility using relative decorrelation time.

3. At high error rates (10%), the APC/SQ algorithm becomes overwhelmed such that, not only are more words wiped out, but the effects spill over into other words. This reduces the average intellibility even further. For LPC-10, this effect appears less severe, perhaps due to the shorter "memory span" of the LPC-10 algorithm.

SECTION 4

LIST OF REFERENCES

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[3] D. Kemp and J. Fussell, "Evaluation of Two Narrowband Speech Algorithms," op. cit., pp. 990-993.

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[7] -, Documentation for LPC-10 Version 43, National Security Agency, January 1981.

[8] -, <u>Documentation for APC/SQ, Version 08: 9600 BPS Adaptive Predictive Coder with Segment Q</u>, National Security Agency, 12 May 1981.

APPENDIX A

LPC SOFTWARE AND TEST EQUIPMENT

A.1 INTRODUCTION

This appendix presents an overview of the LPC-10 speech compression algorithm and describes the use of the algorithm in the end-to-end computer simulation used to evaluate the performance of the algorithms. A brief description of the TRACS receiver model is also included because of its use as the model for the communications link.

A.2 LPC-10 ALGORITHM DESCRIPTION

The LPC-10 algorithm is a bandwidth compression technique used for transmission of narrowband speech. The algorithm digitally samples speech at 8 KSPS and uses a tenth order predictor to compress the output data rate to 2400 bits per second. The algorithm is structured as two functionally independent subsystems, a transmitter or analyzer, and a receiver or synthesizer. The LPC-10 algorithm was originally programmed on a Philco-Ford Signal Processor. A FORTRAN version, designed for use on DEC PDP computers, was provided to MAXIM Technologies by NSA for use in this study.

Figure A-1 shows the signal processing chain for the LPC-10 transmitter. The analog speech signal at the input is bandpass filtered with a gradual roll-off below 100 Hz and a sharp cutoff above 3800 Hz. The signal is then sampled in the A/D converter at 8000 SPS and converted to digital

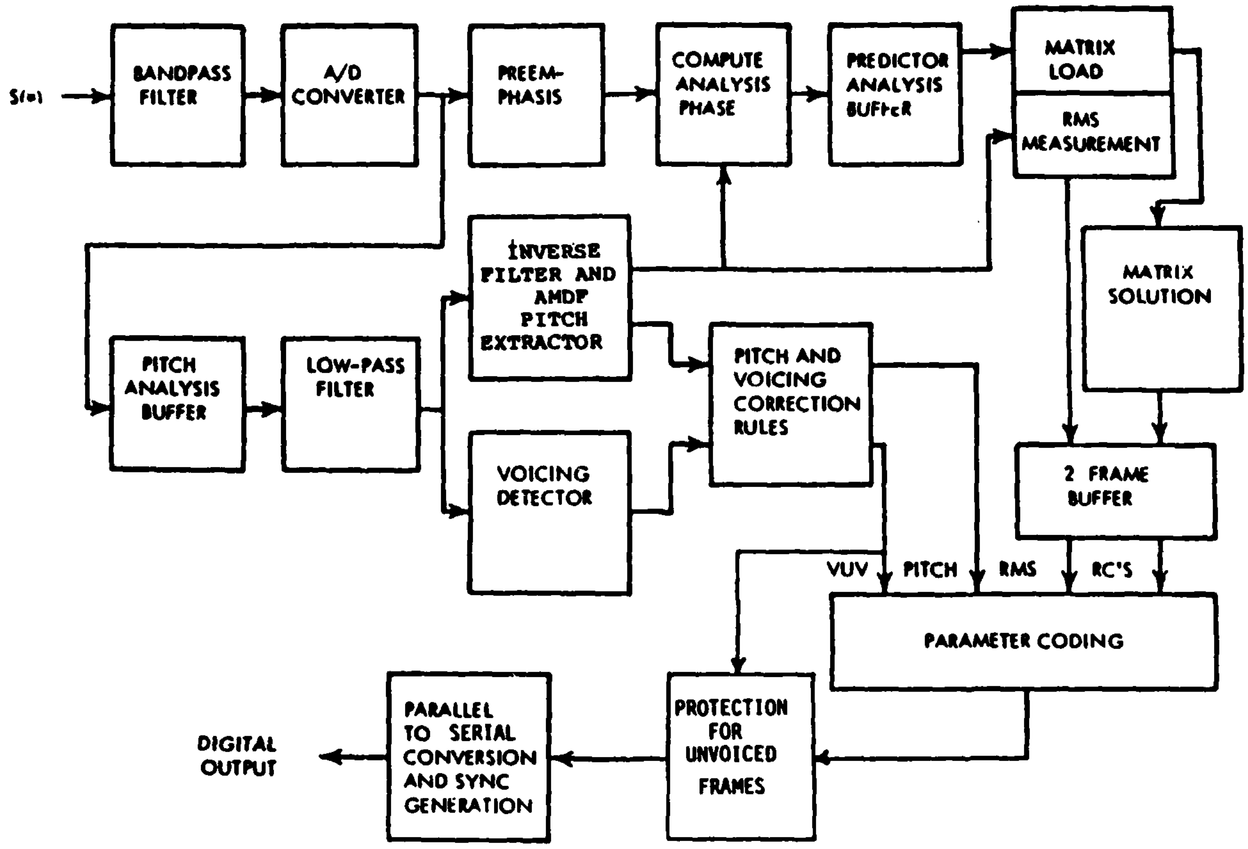

Figure A-1. LPC-10 transmitter.

samples. For pitch extraction, the data is low pass filtered by a fourth order Butterworth filter with a bandwidth of approximately 800 Hz before being fed into the Average Magnitude Difference Function (AMDF) algorithm and, in parallel, into the voicing detector. The voicing detector uses an energy measure with a number of adaptive energy thresholds, zero crossing analysis, and the AMDF max-to-min ratio to make its decision that the data is voiced or unvoiced. A voicing decision is made on each half frame. Pitch and voicing rules apply smoothing and isolated correction to the pitch and voicing values.

The Predictive coding analysis uses a tenth order modified covariance (ATAL) algorithm. Single time-constant digital pre-emphasis with a treble boost above 700 Hz, is applied to the data to be used in the predictor analysis. Parameters are encoded and sent for either voice and pitch or sustained voice input. Fifty-three information bits and one synchronization bits are sent in each frame. A set of five Hamming (8,4) block code words are used to protect the most significant its of these parameters.

The receiver, shown in Figure A-2, decodes the seven bit pitch/voicing word, with the capability of correcting one bit error with respect to voicing. The voicing decision is then smoothed, along with the detection and smoothing of some pitch errors. Unvoiced and transition frame error detection and correction is also done by the Hamming (8,4). Finally, the energy parameter and the reflection coefficients are decoded.

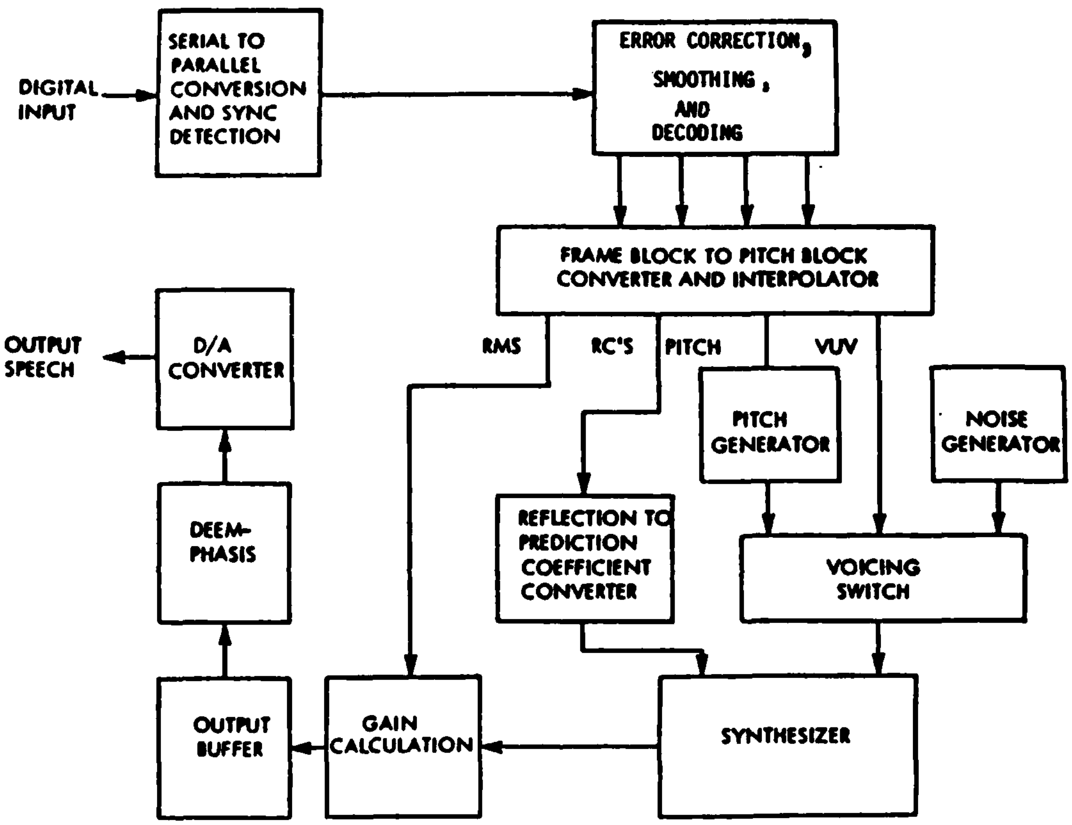

Figure A-2. LPC-10 receiver.

Decoded parameter values are converted from frame block to pitch period block format by interpolation of the pitch, RMS value, and the reflection coefficients. Each particular pitch period is assigned a set of coefficients to which its end is closest in the frame. The number of pitch periods that can fit into the current frame is determined. The conversion produces one or more sets of RMS, pitch, voice/unvoice, and reflection coefficients that are then transmitted to the appropriate function coefficients.

The synthesizer generates one pitch period at a time by use of a direct-form recursive filter with the predictor coeficients as its weights. It operates with a constant excitation signal for voiced frames and a white noise excitation for unvoiced frames. Digital data at the synthesizer output is triple buffered to meet the requirements for weighting pitch periods across frame boundaries. This data is de-emphasized prior to being loaded into the buffer. The output then drives a Digital-to-analog converter after passing through a 3800 Hz low pass filter.

A.3 LPC-10 ANALYSIS SOFTWARE

Figure A-3 shows the signal processing chain used in the analysis. Figure A-4 shows the LPC-10 signal processing chain developed for preparing the actual Phonetic Alphabet Comprehension Tests (PACTs) through use of the error statistics and patterns generated by the various channel and equipment models. To construct the PACT tests, a word extraction program stores 55 frames of LPC voice parameters for each of the 26 alphabet words in the test. These frames of un-distorted data are prerecorded on audio cassette tapes and later played back through an A/D data acquisition system

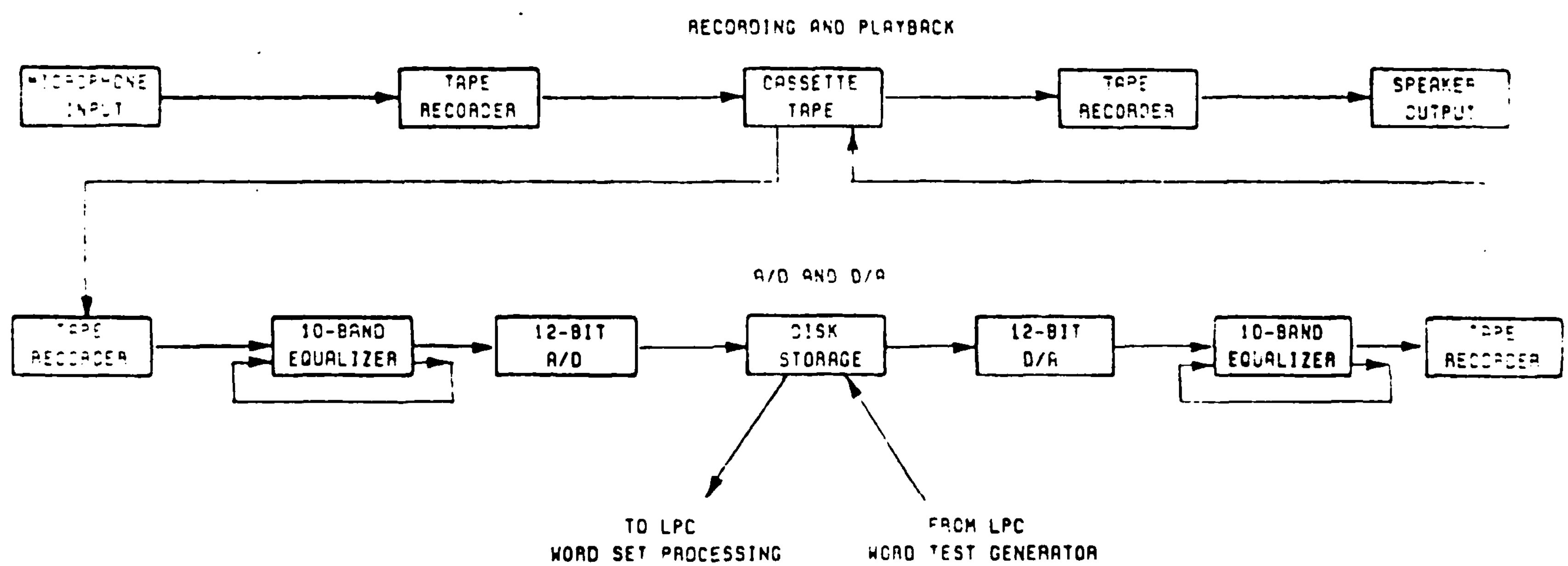

Figure A-3. LPC-10 peripheral processing.

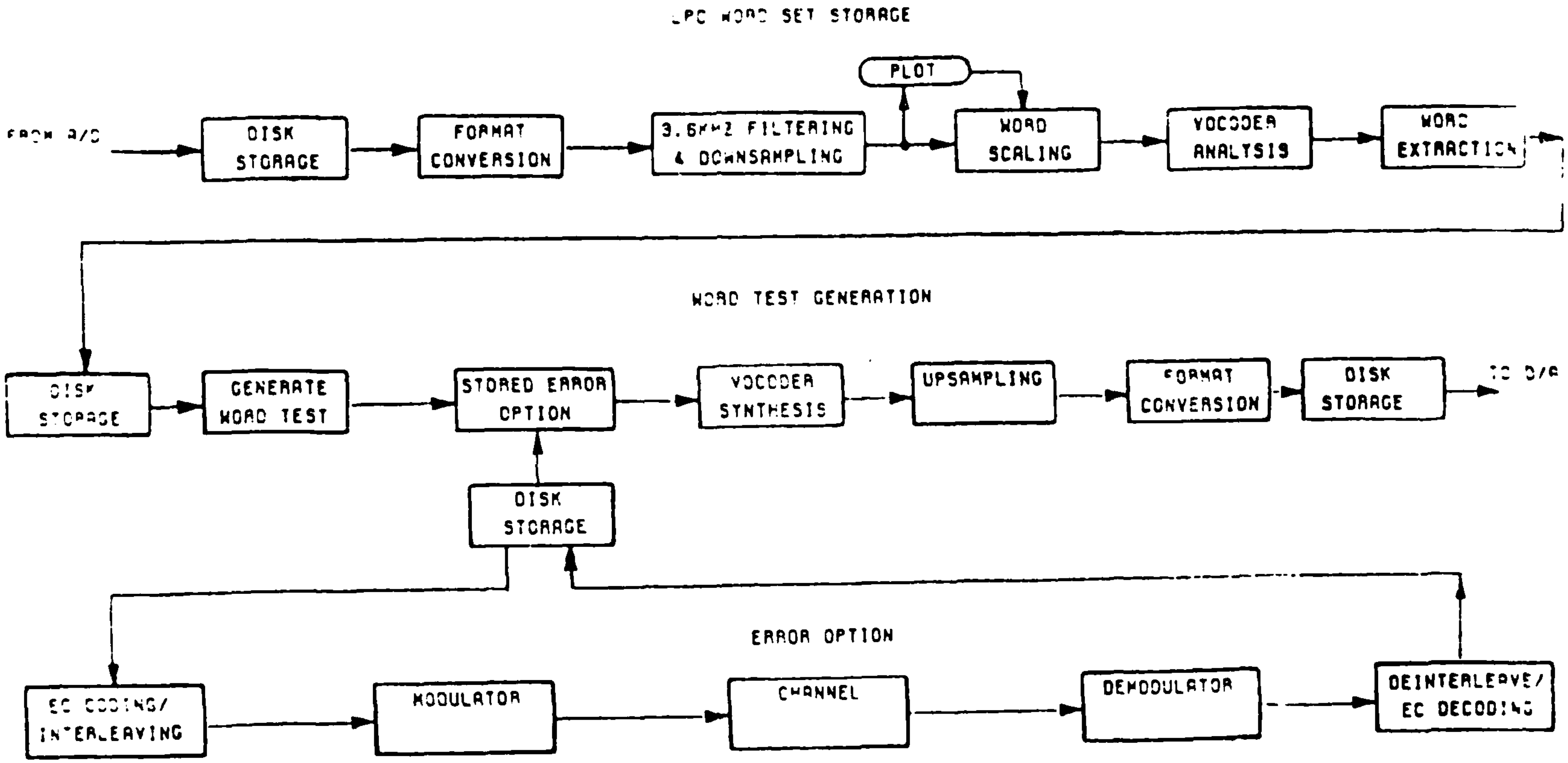

Figure A-4. LPC-10 VAX processing.

for entry into the computer. The audio signal is passed through an audio equalizer twice, where it is bandpass filtered to reduce aliasing, and level adjusted to maximum the dynamic range of the A/D. The digital data is then stored for playback with a variety of fading and noise error patterns.

The top line of Figure A-3 shows that the stored 16 KSPS digitized voice file formats are converted so that they are compatible with existing speech analysis software in the MAXIM Systems Analysis Testbed. In this software, the data is assumed to be low-pass filtered to 3600 Hz and sampled at 8 KSPS. The digital voice is then processed by the LPC-10 analysis section to generate the 2400 BPS data. Bit errors for the fading and non-fading channel errors are inserted into the data. The distorted data is then passed to the LPC-10 (or APC) synthesis program. Finally, the output of the synthesis program is fed to the D/A converter for output recording on an audio recorder for later evaluation by the test evaluators.

APPENDIX B

APC/SQ SOFTWARE

B.1 INTRODUCTION

This appendix presents an overview of the APC/SQ speech compression algorithm and describes the use of the algorithm in the end-to-end computer simulation used to evaluate the performance of the algorithms. The APC/SQ algorithm was originally programmed for use on the Philco-Ford Signal Processor. A FORTRAN version of the software, designed for use on DEC PDP computers, was provided to MAXIM technologies by NSA for use in this study. The only modifications made by MAXIM to the software consisted of the addition of a subroutine to handle data I/O with a tape drive, and the addition of a call in the MAIN program to add the sumulated channel errors to the compressed speech data. This Appendix presents an overview of the algorithm, a detailed description can be found in Reference [8].

B.2 APC/SQ ALGORITHM DESCRIPTION

Adaptive predictive coding is a coding technique in which the feedback signal is composed of the actual linear prediction residual rather than a pulse train or noise as are used in the LPC-10 algorithm. As a result including the prediction residual in the transmission, a higher data rate of 9600 BPS is required for APC/SQ, with an attendant higher speech quality than LPC-10. The linear prediction algorithm for 9600 BPS APC/SQ is composed of both a short-term and a long-term predictor. The short-term predictor is used

similarly to the LPC-10 linear predictor and is used to remove redundancies in the model of the speaker's vocal tract. The long-term predictor is used to remove the effects of speech pitch redundancies. The algorithm is structured as two functionally independent subsystems, the transmitter or analyzer, and the receiver or synthesizer. Figures B-1 and B-2, respectively, describe these two subsystems.

At the input to the transmitter, the Segment Quantization step divides the input speech data into blocks of 190 input time samples that are eventually used to generate 60 coded speech parameter bits and 180 segmented quantization coder signal bits. (Note that with an input sample rate of 7600 SPS, the net output data rate will be 7600 x (180 + 60) V 190 or 9600 BPS.) Following the segmenter, Signal Analysis consists of 4 major segments. Signal conditioning starts by first removing the long term DC bias from the input speech samples. This is then followed by a preemphasis step that applies a trebal boost filter to the speech data. The filter is implemented as:

$$\text{SPEECH}(i) = \text{SPEECH}(i) - \text{SPEECH}(i-1) + 0.5 \times \text{SPEECH}(i-2) \qquad (1)$$

This filters the signal by the inverse of the spectral weighting for average speech. All subsequent processing is performed on the preemphasized waveform.

The TAU and ALPHA speech parameters are next calculated. The pitch value TAU is calculated using the Absolute Magnitude Function (AMDF). This function takes the form:

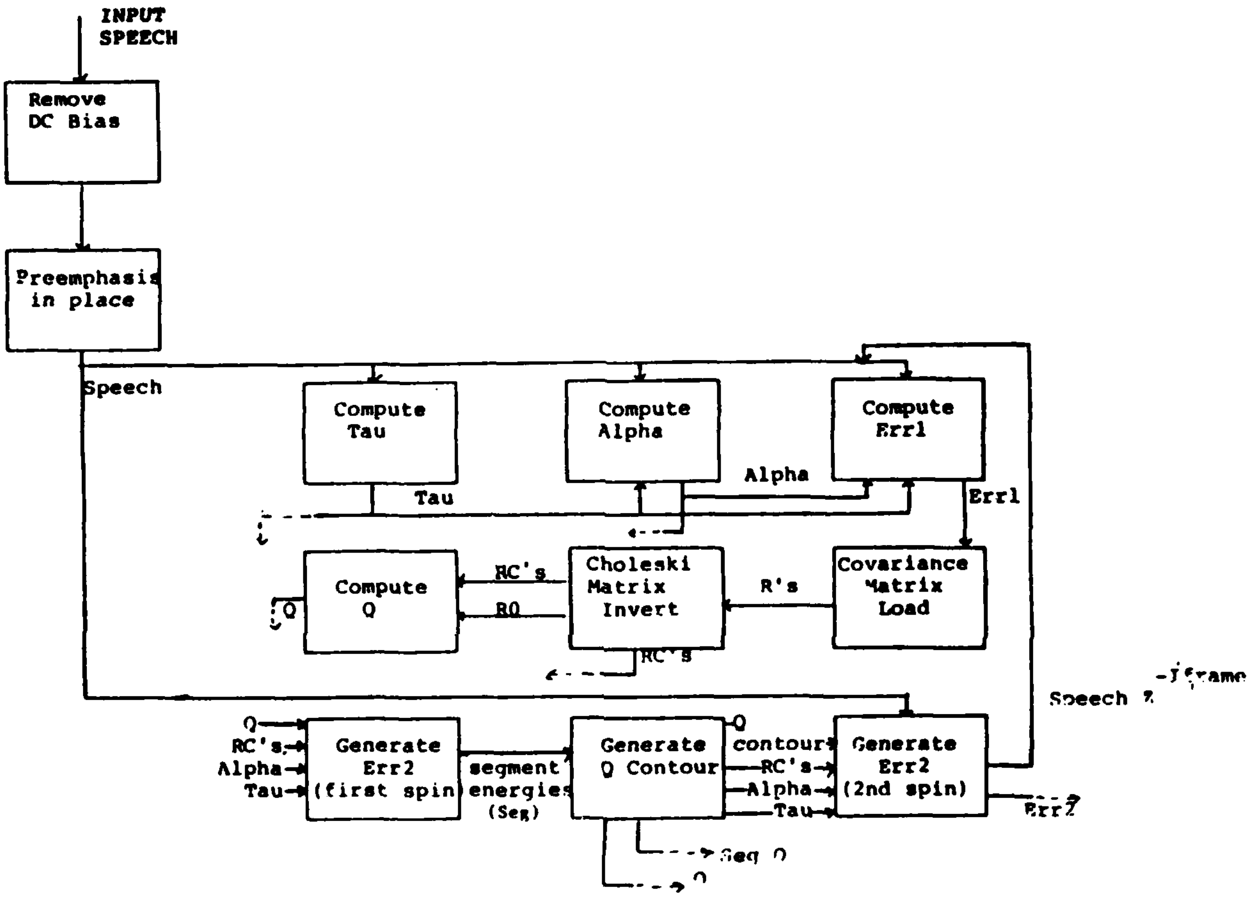

Figure B-1. APC/SQ transmitter/analysis diagram.

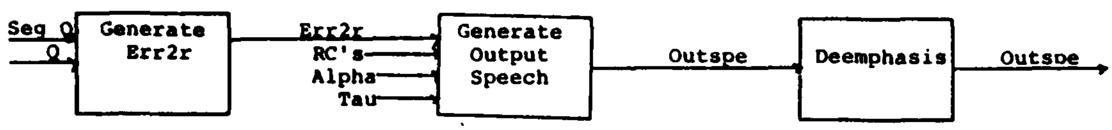

Figure B-2. APC/SQ receiver/synthesis diagram.

$$\mathrm{AMDF}\ (i) = \sum_{K=1}^{N} (\ \mathrm{SPEECH}\ (k) - \mathrm{SPEECH}\ (k-i)\ ) \qquad (2)$$

where i corresponds to the time lag associated with the candidate pitch value. The pitch correlation coefficient ALPHA, is the gain applied in the pitch loop. This value is calculated from the current input speech and previous speech history from the last frame by:

$$\mathrm{ALPHA} = \frac{(\ \mathrm{SPEECH}\ (i)\ \mathrm{SPEECH}\ (i-\mathrm{TAU})\ )}{(\ \mathrm{SPEECH}\ (i-\mathrm{TAU})\ \mathrm{SPEECH}\ (i-\mathrm{TAU})} \qquad (3)$$

Limits are placed on ALPHA to assure stable behavior. The TAU and ALPHA parameters are encoded and used in the computation of the reflection coefficients. The reflection coefficients are calculated from a fourth order covariance matrix. A fourth order Choleski Matrix Invert is performed on the input "err1" signals.

The last functional segment of the transmitter is the generation of the coded 180 segmented quantized signals that along with the 60 coded parameters make up the values transmitted through the channel. Each segment is described by 2 bits determining the level of quantization relative to the transmitted values. Final coding of the parameters includes the use of a (21,16) modified Hamming error correction code of the most critical sensitive parameter bits.

The functional operation of the synthesizer is shown in Figure B-2. Channel errors are applied to every section of the 240 segmented quantized transmitted bits just before the start of the receiver (or synthesizer) section of the APC/SQ code. The receiver unpacks the 240 received bits into parameters and segment quantized values (e.g. residual speech). Error correction is applied to parameter and segmented quantized values. The residual speech is used to excite the long-term and short-term filters to reconstruct the audio speech waveform.

Within the System Analysis Testbed, the APC/SQ software is used in the same manner as the LPC-10 algorithm was tested. Audio signals are pre-digitized at 16K SPS, decimated internally to the required 7600 BPS, corrupted with fading and noise, then synthesized back into audio for evaluation by the listeners.

APPENDIX C

TRACS MODEM DESIGN

C.1 INTRODUCTION

The receiver design selected for use in the analysis simulations was the Transmitted Reference Auxilary Command Signal (TRACS) phase demodulator, shown if Figure C-1. The demodulator was developed as a mitigation design based on the idea that improved signal demodulation is possible if a priori knowledge of the channel is available. Knowledge of the channel phase for such a demodulator is obtained by transmitting a known reference signal (ie. PN sequence, tone or known data value) along with the user data. The amount of the channel caused phase shift can be calculated from the known reference signal. The conjugate of this phase estimate is then used to remove the channel phase shift from the demodulated user data. Corrections in phase may be updated for the received user data on a sample by sample basis, over an entire bit, or averaged over a longer time interval, as required.

The main requirement for any TRACS design is that, while the reference and user data is uncorrelated, the channel maintains frequency coherence (i.e. correlation) over the signal bandwidth of the reference and user data. Channel caused phase variations that occur over the data will than be correlated and approximately the same as the variations that occur over the reference signal. For long channel decorrelation times with wide frequency coherence, and with both signals occupying the same frequency bandwidth,

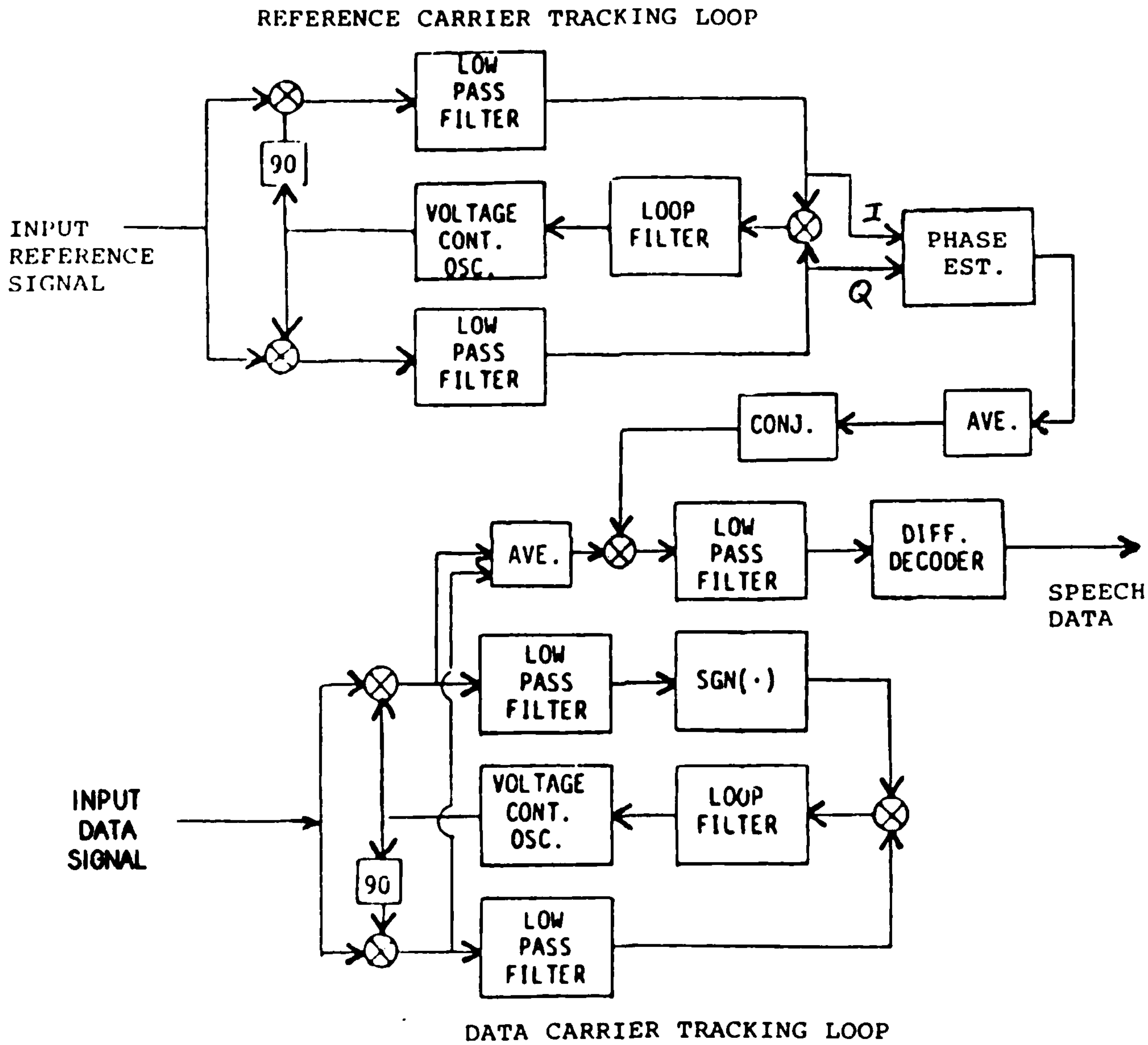

Figure C-1. TRACS receiver model.

the channel caused phase and amplitude signal variations of the two signals will be virtually identical.

The main reason for the improved BER performance of the TRACS demodulator over that observed for conventional PSK demodulators is the removal of the phase error. If the power supplied to the reference signal is instead added to the speech data, the errors due to the channel phase corruption will remain. This result is observed in Figure C-2, where a limit in BER performance is reach^d, even at infinite Eb/No. TRACS demodulation will remove mos of the phaes caused arrors, and bring the BER performance down to the Slow Reighly Limit where errors are a result of amplitude fluxuations only. Therefore, the TRACS design will provide near optimal performance.

C.2 TRACS IMPLEMENTATION

The *only* requirement of the TRACS modem is that an accurate estimate of the channel phase can be made. One means of achieving this is to time multiplex the reference signal with the data signal. The duty cycle time on the reference signal need only be long enough to obtain a reliably strong and accurate signal for a phase estimate at the demodulator. After de-multiplexing, an estimate of the channel phase is obtained from the reference tracking loop, and applied to the user data output from the carrier tracking loop. As an example, if the channel produces a 180 degree phase advance in the data and reference signal, the conjugate of the reference signal multiplied with the data signal will remove the phase advance from the user data. It should be noted, that a TDM will increase the data rate and transmitted signal bandwidth. The amount of increase will be related to

the duty cycle. A 0.5 duty cycle was used for the data links simulated for this analysis.

C.3 TRACS FDM IMPLEMENTATION

Another means of implimenting a TRACS modem is in FDM systems where users or groups of channels are multiplexed and hopped around a defined frequency bandwidth. TRACS will provided a valid channel estimate as long as the channel remains coherent over the entire group bandwidth, or if the user data channel remains close enough to the reference signal channel during all hops.

For example, if a system can support twenty channels per group, and possibly several users per channel, a TRACS demodulation design approach would require that only one of the present channels contain a known reference signal. This signal reference would be used by TRACS demodulators on any or all of the remaining channels for added performance in the fading channel. Standard demodulation would be performed by those remaining non-TRACS users receiving data on the same channel, or communicating on other channels, without any need or use of the reference signal. The fact that TRACS implementation requires no change to existing receivers or the a satilite FDM system, provides a cost effective means of upgrading a system to a survivable mode of operation.

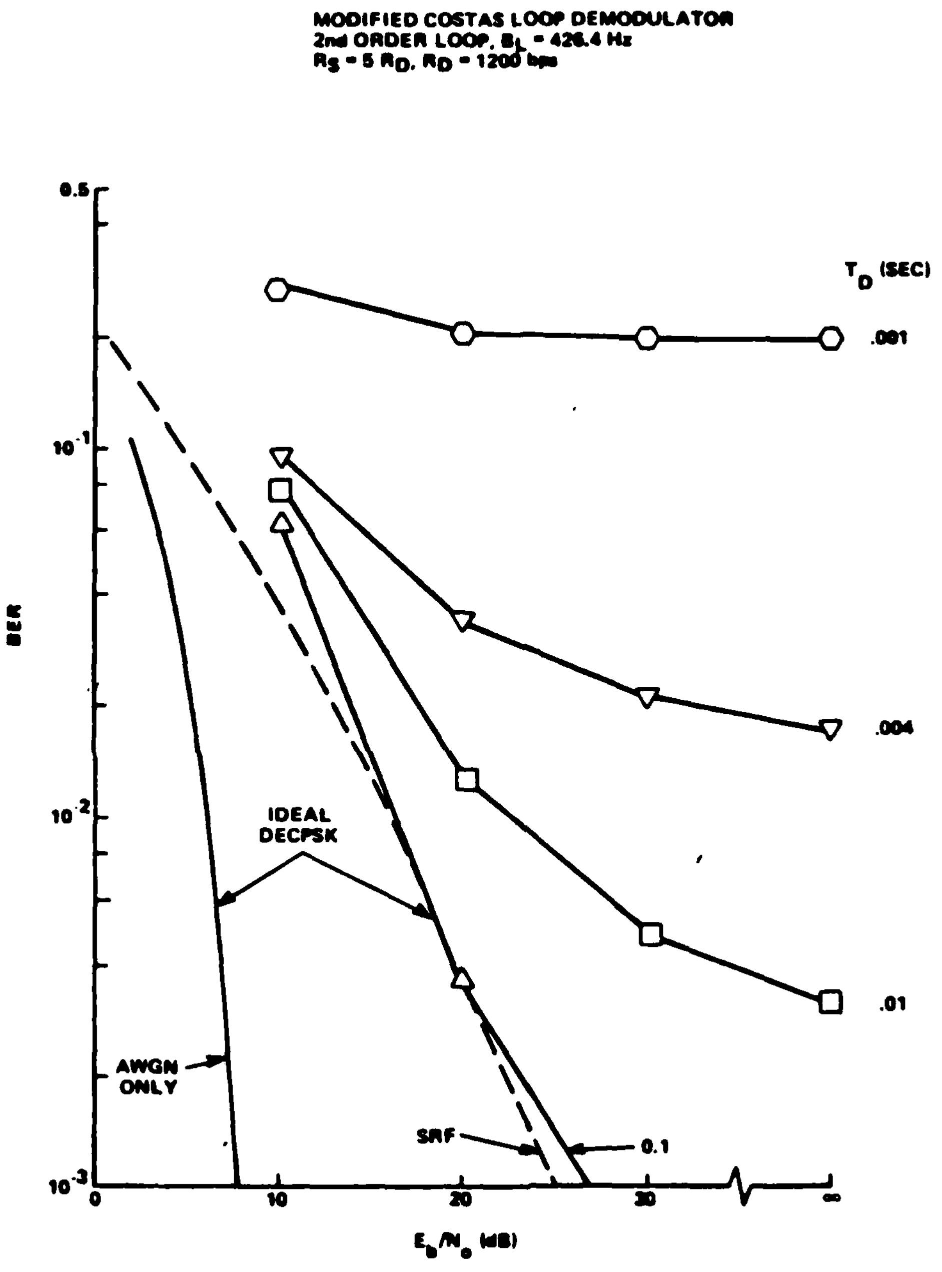

Figure C-2. DECPSK BER performance.

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